ELECTRONICS, MUSIC AND COMPUTERS

by

Alan Conway Ashton

December 1971

UTEC-CSc-71-117

ACKNOWLEDGEMENTS

I wish to thank Professor Robert S. Barton for his motivation, stimulation and ideas in special-purpose systems design, information representation and programming structures. I appreciate the patience and encouragement of Professors David C. Evans and William J. Viavant during my graduate study.

I am particularly indebted to Robert Bennion for many design suggestions, interesting discussions and the hardware implementation of the prototype equipment.

INTRODUCTION

Electronic and Computer Technology has had and will continue to have a marked effect in the field of music. Through the years scientists, engineers, and musicians have applied available technology to new musical instruments, innovative musical sound production, sound analysis, and musicology. At the University of Utah we have designed and are implementing a communication network involving an electronic organ and a small computer to provide a tool to be used in music performance, the learning of music theory, the investigation of music notation, the composition of music, the perception of music, and the printing of music.

The computer-aided music tool is shown in figure 1. The computer serves as a communications device that aids the user by performing several functions such as the storage of information, performance of pre-assigned sequences, and retrieving of specific stored data. The computer receives input from a typewriter keyboard, a light pen attached to the display scope, the organ manuals, pedals, and stop settings. In turn, the computer controls the audio tone production, lights which indicate the organ keys, the display scope and a color generator.

There are many interesting things which can be done with this hybrid configuration; for instance, computer assisted learning of music can be investigated, experiments of sensory perception may be carried out, keyboard performance can be studied and new sounds, rhythms and melodies may be intricately interwoven. I will discuss these and other possibilities further after I describe the computer-aided music tool fully (chapter 4). But first, it will be instructive to look at some acoustic properties of sound, the development of electronic musical instruments and the use of computers in the field of music so that in the overall picture the value and position of our system can be assessed and evaluated. Finally, I will look at recent trends and speculate on the future of electronics and computers in the field of music.

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ABSTRACT*

The uses of electronic and computer technology in music are surveyed and a computer-aided music tool is described which consists of a small computer, electronic organ, and a line-drawing display scope. The computer receives input from a teletypewriter keyboard, the organ keys and stop tablets. The input information is processed according to stored programs, and output is directed to the organ tone generators, filter selection switches, lights which designate the manual keys, volume control and the display scope. Thus, the organ is full-duplexed through the computer, since the keys are not directly connected to the tone generators, but are linked via the computer. A basic internal representation of stored music is developed which consists of an interpreted file containing a sequence of variable length commands and typed data. Information transfer rates of a secondary storage tape cassette recorder is adequate for most classical organ selections, which may be directly entered into the computer by performance on the manual keyboards and pedals, or keyed-in, note by note, either from another type keyboard or from the organ keys themselves.

The interactive organ-computer communication network allows real time visual, aural and written response. This makes the musicational

*This report reproduces a thesis of the same title submitted to the Department of Electrical Engineering, Division of Computer Science, University of Utah, in partial fulfillment of the requirements for the degree of Doctor of Philosophy.

tool, as it is called, useful for programmed music courses, computer aided instruction of music theory, harmony and keyboard performance, and conducting human information processing experiments in the investigation of multisensory perception.

The methods of music production, processing power, and capabilities of the musicational tool are compared to the techniques of other electronic and computer-aided musical instruments and the influence of electronic and computer technology in music of the future is discussed.

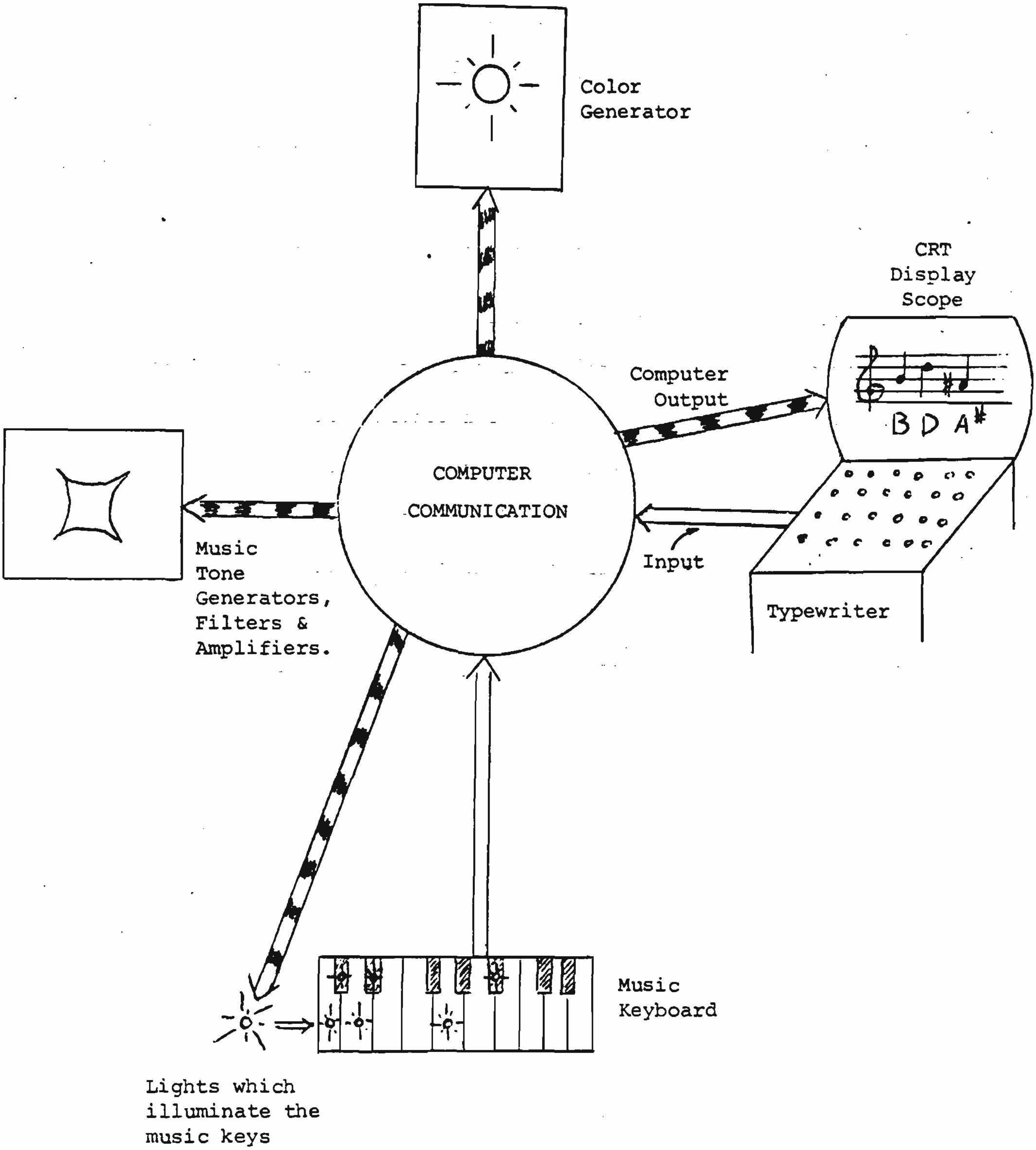


Figure 1

The Musicational Tool

CHAPTER 1

MUSIC ENGINEERING

The auditory sensation sound is produced by the impact on the eardrum of a pressure wave propagated in an elastic medium. Thus, sound results when air or some other medium is set into motion. The frequency of a sound wave is the number of recurrent waves passing a certain observation point per second. The physiological sensation depending mostly on frequency is pitch. Auditory perception of the intensity or energy contained in a sound wave is the loudness of sound. In other terms, the volume of sound is the physiological counterpart to the amplitude of the wave.

Sounds may be organized into five main categories [71].

1) "Pure" tones,
2) ordinary musical tones,
3) clangorous sounds,
4) ordinary noises, and
5) white noise.

The first category consists of pure sine waves which can be represented as a function of time as $s(t) = A\sin(2\pi ft + y)$ where A is the amplitude, f the frequency and y the phase shift of the sine wave. The sound of a sine wave to a listener is dull and becomes tiresome after a while due to the exact repetition of the waveform. The second group of sounds in their steady state condition may be represented as a mixture of sine tones represented by a Fourier Series: $S(t) = \sum_{n=1}^{\infty} A_n\sin(2\pi nft + y_n)$. Two interesting waveforms in this group are square waves and sawtooth waves. A square wave contains only odd harmonics and can be represented as $S(t) = \sum_{n=1}^{\infty} \frac{A1}{2n-1}\sin[(2n-1)2\pi ft]$. The result of a square wave is a hollow sound, much like that of a clarinet. A sawtooth wave contains harmonics whose amplitudes are inversely proportional to their frequencies and is represented as $S(t) = \sum_{n=1}^{n} \frac{A1}{n}\sin 2n\pi ft$. A sawtooth wave produces a buzzing-like sound not very dissimilar from that of an English Horn. The third category of sounds consists of those having inharmonic partials and include sounds produced from bells, gongs and cymbals. The next group is comprised of any kinds of sounds occurring contiguously, and the final category is composed of sizzling, hissing and howling noises in which all audible frequencies occur at random times.

Music is the auditory perception of sounds which have been organized in some meaningful way. Electronic music is that music which is produced, modified or combined with the aid of electronic equipment. The organizational procedure of electronic music can be divided into six highly-interacting areas: Waveform generation, modification, coordination, storage, control and notation.

Waveform Generation.

Generation of sounds can be done with any device capable of producing undulating signals, whether they be in a vibrating medium, an electrical circuit or in the air. The electronic musical instruments described in later chapters use variations of the following five types of tone sources: 1) Purely electrical, 2) rotational scanning,

3) vibratory, 4) electrical pickup of concrete sounds, 5) digital. Purely electrical generators include oscillating arcs, relaxation oscillators, vacuum tube circuits with feedback, thyratron oscillators, condenser-inductance discharge, radio-frequency beat systems and transistor oscillator circuits. Electrical circuits employing an amplifier are made to oscillate by feeding back a portion of the output voltage in a certain phase relationship into the input bias to overcome the attenuation losses of the circuit and thus effect the oscillation. Very stable vacuum tube oscillators were hard to design because of the fluctuating characteristics of the tubes with time. Thermal effects in the tubes as well as variations in anode voltages and cathode currents had drastic effects in changing the constants of a tuned circuit. The first tube or valve oscillators to generate a sine waveform were especially unstable. One of the popular vacuum tube oscillators finally developed was the Hartley Oscillator which had fairly good stability.

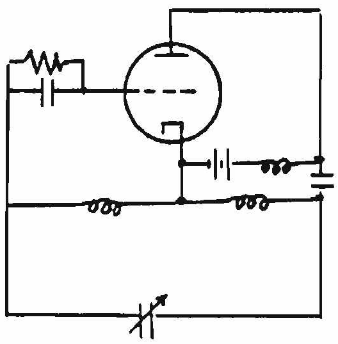


Figure 1-1
Hartley Oscillator

Most of the popular electronic organ manufacturers including Allen, Baldwin, Conn, Electrovoice, Gulbransen, and Lowrey used this oscillator or a modified version thereof in their tone generators. To maintain the degree of stability needed, a coil inductance having a fairly high Q was used in conjunction with the Hartley. Versions of this oscillator produced sine, sawtooth, and pulse waveforms. A very useful waveform is the sawtooth since it contains essentially all the harmonics of the fundamental. Both vacuum tubes and gas tubes were used in sawtooth generating circuits. One of the simplest relaxation configurations used a neon tube.

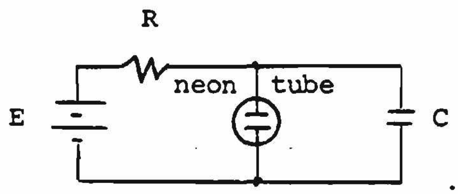


Figure 1-2
Neon Tube Oscillator

The neon tube discharges when a certain potential is applied across its electrodes, and it remains conducting until the voltage is reduced below a certain cutoff point. The waveform is produced by the capacitor charging up exponentially until the tube ignition voltage is reached, which discharges the capacitor almost immediately. The frequency is determined by the values of E, C and R. One of the problems with neon tubes is that their firing characteristics change with time. Usually the tubes had to be "broken in" until the responses became somewhat stabilized. Neon tubes have also been employed in frequency division circuits where every other incoming pulse triggers the next stage. These circuits are designed so that the oscillation frequency of each stage is a little slower than half the frequency of the preceding stage. When the first pulse comes in, the capacitor in parallel with the neon tube charges extra but not enough to fire the tube. By the time the next pulse hits, the capacitor has nearly reached the ignition voltage and the extra surge of voltage triggers the tube, and the process starts over. Multivibrators have also been used extensively in frequency dividing networks.

With the availability of cheap transistors, most present day oscillators and dividers are built from transistors. When transistors were first developed and attempts were made to use them to replace tubes, much trouble was encountered because of the high power dissipation and low temperature breakdown of germanium. Silicon transistors, manifesting a higher resistance to heat, solved this problem. The obvious advantages of transistors are long life, low voltage requirements, little risk of hum pickup, and compact size. Integrated circuits (ic's) are now being used in some organ designs, but they are still a little more expensive than discrete components for tone generators and divider circuits. The ic's certainly provide more convenient and compact packaging, and they facilitate much easier debugging. When a failure occurs, the ic package can be replaced as opposed to the testing of many capacitors, resistors and transistors to find the trouble. Semiconductor manufacturers are now producing specialized integrated circuits for the electronic organ industry [117, 96, 113]. Flip flops, for instance, are used frequently in oscillator and frequency divider circuits.

An interesting way of producing weird sound effects is with the beating, or heterodyning of two radio frequency oscillators. The high frequency oscillations are above the audio range, but when two differing frequencies are mixed, a resultant beat occurs which may be in the audio range. The sound effects are created by rapidly changing the frequency of one of the two r.f. oscillators.

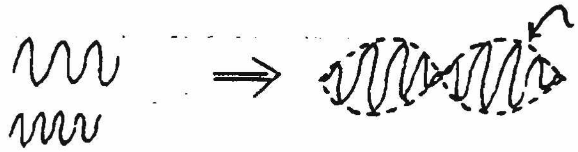


Figure 1-3
The Heterodyning of Two Signals

Electrical generation of oscillating signals is just one of the many ways which have been used. Rotary devices with scanning pickups were used extensively in many electronic music instruments. The types of pickup used were magnetic, electrostatic and photoelectric. The acoustic siren and the phonograph, of course, operate upon this principle also. Of the many rotating devices, only the Hammond electromagnetic pickup succeeded commercially. In these systems an irregular shaped rotating conductor alters the magnetic field in a permanent magnet and induces a current in a coil wound around the magnet. The varying current is then amplified and fed to the rest of the circuitry.

In designing electromagnetic generators the effect of hysteresis had to be considered. In some designs the lag was helpful in obtaining complex waveforms. Rotating electrostatic pickup was accomplished by having two discs facing each other, one of which rotated. Voltage was impressed across the two discs and as one rotated the varying capacitance set up an oscillating voltage in the circuit. One disc contained metal etchings of desired waveforms, while the other had the scanning pickup. In some schemes the disc containing the waveforms rotated and in other instances the scanning plate revolved. As the area between the scanner and waveform varies, so does the capacitance. The third type of rotating generator was photoelectric. One of the simplest photoelectric generators consisted of a wheel with a series of holes in it which allowed a pulsating beam of light to fall on a photoelectric tube. Photoelectric cells can produce a current proportional to the intensity of light falling on the cell, and the varying current becomes the electrical signal. In more sophisticated designs, a waveform is etched on the disc, and as it rotates variable amounts of light appear at the photocell. Sound tracks on movie films are an example of this technique. Rotating drums and records have also been used with the different pickup methods.

Vibrators make up a third class of tone generators. There are so many combinations involving vibrating elements, exitation methods and pickup schemes that it is probably easiest to see three separate tables and imagine how each of the vibrators can be put into vibration by each of the exciters, etc. [102].

Vibrating Elements	Exitation	Pickup
Wires, reeds, strings, bells, metal plates, tubes, bars, tuning forks, sounding boards, membranes, springs	hitting, plucking, blowing, bowing, twisting, resonance, magnetic forces	electrostatic, electromagnetic, photoelectric, microphonic, piezoelectric, interrupted contact, modulated resistance, thermoelectric

Concrete sounds, or those sounds made by ordinary physical objects, are a source for many electronic music enthusiasts working with classical tape studio equipment. The sounds range from those of a howling coyote, an ambulance siren, and downtown traffic to the mooing of a cow, the creaking of a door and the dripping of a leaky faucet. These sounds are generally recorded on tape by microphone and then modified by the various techniques of the sound modification category.

The digital computer has opened up whole new areas of sound generation and waveform processing. Digital sound generators consist of programmed routines which produce lists of numbers which are transformed to analog waveforms by a digital-to-analog (D/A) converter and a low pass filter. The numbers generated by the computer are samples of the waveform at certain time intervals. The number of samples per second determine the highest frequency possible in the sound generated. It turns out that the number of samples must be 2n/sec where n is the highest frequency. The digital samples are converted to voltage pulses by D/A converters and these pulses are smoothed out by the filter to produce the final analog signal.

The generation of say 30,000 samples per second for a frequency response of 15,000 cycles per second takes some time on the computer, especially if the synthesized waveform is complex. Usually the waveform samples are computed and stored in an array to be later output to the D/A equipment. Anywhere from 2 seconds to several seconds may be required to compute one seconds worth of sound. The trend is now to store digital samples of one period of certain cyclic waveforms in tables in the computer. A limited number of samples are stored, and if a sample value is needed for a time not represented in the table, simple interpolation is done. The stored waveforms can be combined in various ways as I shall explain later and the resultant sound can be produced in real time if a low frequency response is permitted.

Waveform Modification.

The second category of electronic music production is the modification of the generated waveform. There are two major ways to produce distinctive waveforms which may be further modified by other means. The two methods are additive and subtractive synthesis. Additive synthesis consists of mixing pure tones (sine waves) together to produce complex waveforms and subtractive synthesis consists of filtering out certain frequencies from a complex waveform already containing a large spectrum of frequencies. In additive synthesis, the fundamental is combined with various harmonics, partials and overtones having different amplitudes to synthesize desired sounds.

Analysis of steady state musical tones have shown that distinctive tone colors manifest a certain harmonic content. For example, a flute sound consists of a dominant fundamental with a little bit of 2nd and 3rd harmonics. Violin sounds on the other hand have large contributions from all of the first eight or nine harmonics, with emphasis on the fourth and fifth. Harmonics are multiples of the fundamental frequency. In some electronic organs where additive synthesis is done (Hammond, for example) the tones used to form the result are equitempered and thus the introduced harmonics are not exact multiples of the fundamental. The subsequent sound produced is distinctive and sounds artificial. Additive synthesis is used in some electronic organs to produce chiff, the starting transients accompanying the speaking of an organ pipe. Here at the outset of the tone for a few milliseconds, certain high partials and usually a bit of noise are added to the tone. Very realistic imitations of actual organ pipe sounds have been developed by a number of electronic organ manufacturers, primarily Saville, Allen and Rodgers.

The other major technique of tone coloring is subtractive synthesis. In this technique the filtering is done in a variety of ways, for instance, resonance and formant filtering, high-, low- and band-pass filtering and band exclusion. An example of resonant filtering is the distinctive sounds produced by differently shaped organ pipes. Here the natural resonance of the pipes impart an emphasis to certain frequencies and thus color the tone. Variously contoured speakers have been used in some electronic musical instruments to alter the tone quality of the music produced.

Formant filtering is the electrical counterpart of resonance filtering. The formant theory was developed by Hermann in Germany. This theory was that vocal timbres are produced by the effects of the throat, mouth and nasal cavities on the vocal pitch produced by the vibrating vocal chords. The pulsating air stream from the lungs through the vocal chords set the several resonant cavities into damped oscillation, which may be inharmonic to the frequency of the vocal chords. Hermann called these effects formants [31]. Electrical formant filters are resonant circuits which impart an emphasis region to certain frequencies. These filters only modify the existing harmonics already in the waveform; they can't completely eliminate any or add any. The nature of most sounds depends not only on the harmonic content, but also upon the spectrum or frequency response in time. Formant filters are generally applied to complex waveforms such as sawtooths which have rich harmonic content.

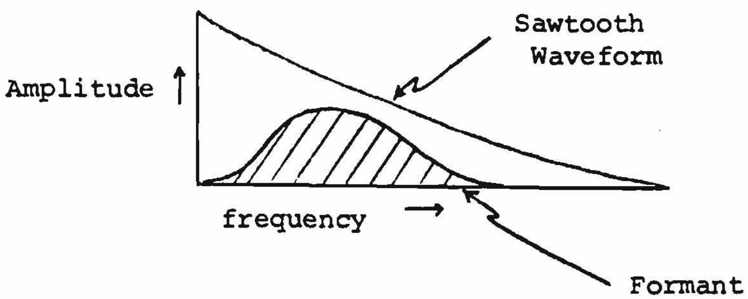


Figure 1-4
Formant Filtering

Formants then, as shown in Figure 1-4, are frequency ranges in which harmonic components are predominant relative to neighboring harmonics. A particular tone quality may have several peaks, or formants along its frequency response curve. Formant filters usually consist of a passive band pass filter with an amplifier with positive feedback. When the gain exceeds the attenuation in the circuit, it alternates at a resonant frequency. Waveforms sent through such a circuit receive a reinforcement of those frequencies at which the filter is oscillating. Passive filters are extensively used to impart tone colors to complex tones. For example, a low pass filter produces an organ flute sound composed of mostly low harmonics. The general classes of passive filters are depicted in Figure 1-5.

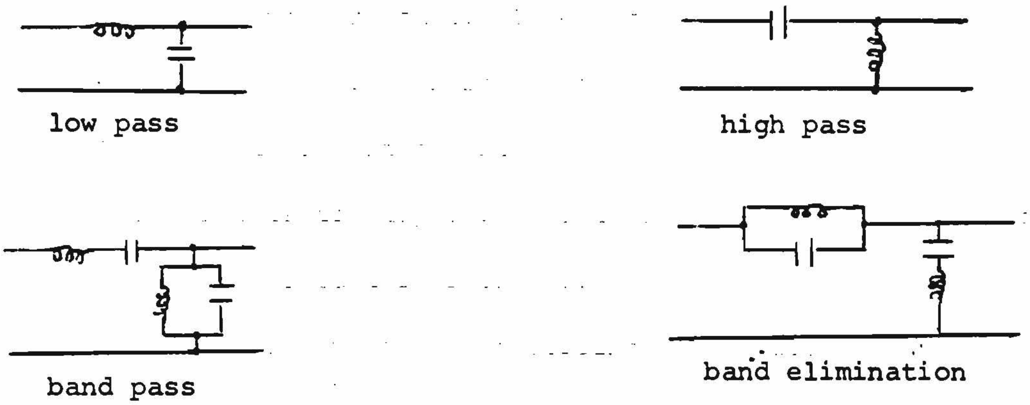


Figure 1-5
Passive Filters

A passive filter is one which contains only resistors, capacitors and inductors, whereas an active filter contains an amplifier.

Some organ manufacturers are now using active filters, which may be made to resonate along with passive filters to imitate pipe organ sounds. The problem with band-pass and resonant circuits is that they filter low notes and high notes differently. The trend now is to have many band responsive filters for a particular tone color over the entire compass of the pitches.

The rate of attack and length of decay of a sound are important parameters which determine tone quality. A violin tone with a percussive attack and gradual decay sounds like a piano, and a piano tone given a time envelope of a violin tone sounds like a violin. Most instruments have exponential attacks and decays as shown in Figure 1-6.

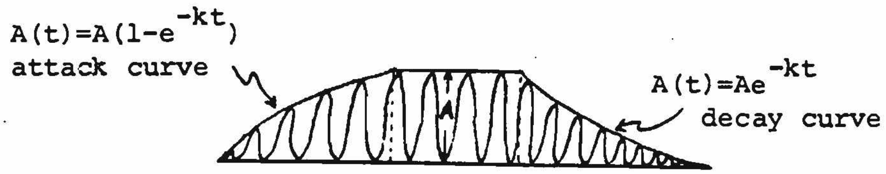


Figure 1-6

Exponential Attack and Decay

Electronic music instruments simulate attack and decay by resistor-capacitor (R-C) circuits in which the charging and discharging time of a capacitor is dependent upon the values of the capacitor and the resistor. Percussive sounds have been imitated very well with electronic circuits having sharp attack times and long decays. There are many commercial electronic band boxes which contain percussion sounds ranging from a bass drum to a cymbal crash. Interesting sounds are obtained by applying a percussive formant to different waveforms, including noise. The gradual attack and onset transients are very important in determining certain Baroque pipe organ sounds. As mentioned earlier, these effects have been simulated very well.

Electronic devices which impart time envelopes to any waveshapes put through them also control the shape of the attack and decay curves as well as the lengths. The computer generation of time envelopes is especially versatile since any curve can be defined as the attack and decay. For instance, linear attacks and decays impart a sciencefiction character to the sounds; whereas exponential curves give the sounds an air of familiarity. People who produce electronic music by magnetic tape manipulation often cut out the portion of the sound containing the attack. Sometimes very interesting and unusual results occur.

Another favorite technique is to splice the attack or decay of one sound onto another sound. Tape recorders are used in a variety of ways to modify sounds. For instance, a noise can be recorded at one speed and played back at another, and this can be repeated *ad infinitum*. A popular trick is to play a recorded sound backwards either at the recorded speed, or faster or slower. The pick up volume can be continually varied by hand as recording takes place, or the tape can be made to accelerate or decelerate quickly by yanking it one way or another. The feeding back of an output signal to the recording head produces a type of reverberation if the returned signal is attenuated. Sometimes, however, the feedback is amplified and the result is a shrieking howl like that produced in a microphone speaker feedback loop. If you have ever wondered why some "electronic music" sounds the way it does you only need imagine some of the above-mentioned shenanigans taking place. It must be conceded, however, that out of some of the bizarre, sound-distortion techniques much interesting music has emerged.

There are other ways besides tape recorder playback loops in which artificial reverberation is created. Reverberation is the repeated echoing of a sound at subdued levels. It is the effect of sound waves arriving at a listener's ear at different times due to environment in which the sound is produced and heard. For example, in an auditorium where live music is performed the sound waves propagate in many directions and are reflected and absorbed by the surrounding objects and finally reach the listener at different times and at various intensities. This overall effect is called reverberation, and it gives sound the quality of "bigness." Thus it is interesting to simulate reverberation artificially.

There are five basic delay techniques used to produce reverberation. They are acoustic delay lines, magnetostrictive delay lines, electronic delay networks, magnetic tape loops and digital delays. Acoustic delays are produced by the finite velocity of sound energy in air or other elastic medium. The air chamber may range from a hollow tube to a large room or auditorium. The usual procedure is to place a loudspeaker at one end of the chamber and several microphones at different points to pick up various delayed sound signals, which are mixed together to produce the reverberated signal. The propagations of vibrations in spring coils and metal plates also constitute acoustic delay lines. The electrical signal is converted by a transducer to a mechanical vibration which is induced in the vibrating coil or plate. Variously placed transducers reconvert the vibrational energy to electrical signals which are combined to effect the reverberated signal. Acoustical delay lines have the drawback that they distort the signal appreciably, especially in terms of frequency response.

Magnetostrictive delay lines are formed from magnetostrictive materials which have the property that a magnetic field applied to one end of the material induces a wave that propagates to the other end [125]. The wave propagation may be either longitudinal or torsional. The advantage of this type of signal propagation is that very little distortion of the wave results, even with a limited number of intermediate signal taps along the wire. To achieve these results with longitudinal propagation, however, the wire or bar must be straight; otherwise, serious phase distortion results. Torsional wave propagation provides longer delays per length of wire and the wire may also be coiled without affecting the phase of the signal, so high frequencies may be propagated without distortion.

The third type of delay line is purely electronic and is produced by cascading resistor, inductance and capacitance delay circuits. The only problem with this type of delay is that some 300 circuits are required to give a delay of 50 ms., thus the amount of hardware becomes prohibitive.

Magnetic tape loops are capable of producing any amount of delay by the placement of several playback heads with feedback at different intervals from the recording head. Schober Organ Company advertises their Reverbatape, consisting of three playback heads, as being capable of producing the sound of a large auditorium to that of a small hall. A good tape reverberator can produce acoustical effects on some sounds indistinguishable from the real thing. The advantage of tape reverberation is that the length of the delays and attenuation of the signals are controllable and it can have full audio frequency response with low distortion. One problem with the Reverbatape is that short, staccato notes are "bounced" back repeatedly instead of being smoothly attenuated.

The final method of reverberation is digital. The processing of digital waveforms in a computer facilitates the simulation of reverberation by programmable control. Mathematical models with iterative techniques (analogous to feedback loops) have been programmed to produce digital reverberation at the University of Illinois [66]. The trouble with digital reverberation is that it is not done in real time. The analog signal must be converted to digital samples, be processed and reconverted to analog. A good system of reverberation at the University of Illinois Experimental Music Studio uses a combination of both magnetostrictive delay lines and tape loops [125].

Vibrato is another modification which can be applied to give warmth to certain types of music. Vibrato is a slow undulation of pitch and amplitude of a sound at about five to nine times a second. A trumpet player produces vibrato by slowly varying the tension of his lips or by slightly altering the pressure of the mouthpiece on his lips. Both of these actions cause variation in air pressure and thus a slight wavering of tone. In a similar manner pulsating variations of air pressure cause pipe organ vibrato. In these cases a variation of both pitch and amplitude cause the vibrato. A violin player utilizes mainly pitch variation as he slides his finger back and forth to produce vibrato.

Some electronic organs vary the gain of an amplifier to produce a vibrato which is really an amplitude modulation. A true vibrato consists of both frequency and amplitude modulation, with the emphasis being on frequency variation. Periodic amplitude variation is correctly termed tremolo or tremulant and is used instead of vibrato on some electronic musical instruments. Tremolo is produced very easily in electronic instruments by adding a slowly oscillating signal to the bias of an amplifier. In photoelectric instruments tremolo is caused by varying the amount of light falling on a photocell by moving a graded transparent slide back and forth in the light beam. The volume of capacitive pickup generators can be varied by increasing and decreasing the distance between the plates. Tremolo is fairly easy to produce, but vibrato is a little more difficult.

In many electronic generators, a slow oscillation is applied to the tone generators to produce vibrato. This works pretty well for individual oscillators, but in a system with frequency dividers the vibrato does not propagate through. In the case of many individual tone generators several rates of vibrato should be used for different pitch ranges. A normal vibrato, when applied to a low note of, say, less than 100 cycles per second, sounds bad because the percentage of variation on low notes represents a significant change in pitch. Generally, vibrato is not imparted to the low pedal notes of an organ, but is saved for the manual tones. Higher pitched notes usually need a wider and faster vibrato than the lower ones.

Vibrato has been produced in a number of ingenious ways on the variety of electronic musical instruments. In those cases where mechanical rotating elements are used, a slight variation in speed produces a good vibrato. An easy way to cause this variation is to use eccentric pulleys on the belt drive as depicted in Figure 1-7. The

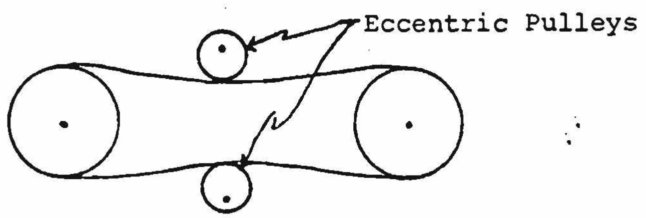


Figure 1-7
Vibrato Applied to Rotating Generators

motion of the outside wheels is back and forth in opposite directions, against and away from the belt drive. In rotating capacitance pick-up generators with opposing disks, the stator can be slightly jiggled to and fro to produce vibrato.

An interesting way of producing vibrato electrically was done by Hammond with a series of time delay networks. The signal is tapped off at different places along the delay line, and the phase differences of the combined signals produces a vibrato. Schober Organ Company uses a phase shift network to electrically create vibrato. Phase shifting produces both amplitude and frequency modulation, similar to the doppler effect. Usually vibrato is applied only at one place for all the notes or else the same vibrato oscillator is used to create all the vibration. In both these cases the tones all vibrate together and are locked in phase. The result is an unnatural sound without much warmth and depth. Some organ manufacturers have overcome this problem by applying different rates of vibrato to various tones which make up the final sound.

Many of the monophonic or melodic instruments were equipped with touch control vibrato. For example, the Trautonium had a liquid resistor in the form of a long tube which was depressed by the keys to form gradations of volume. The lateral movement of the dummy rubber key determined the pitch, so slight undulating movements of the fingers on the keys of the Trautonium produced a very controllable vibrato. Of course, the most sophisticated and precisely controlled vibrato can be programmed and produced digitally, but additional time must be sacrificed since each wave form sample must be modified.

A standard piece of equipment for many organ manufacturers is a rotating speaker which produces a vibrato effect. The large woofer usually is held fixed while the smaller middle and high range speakers are rotated. The vibrato is a result of the doppler effect which is that the apparent frequency of a sound is increased as the sound source approaches the listener, and the pitch drops as the sound source moves away. A variation of rotating speakers is rotating baffles into which the sound is directed. Here again the virtual source of the sound is varied as the rotating baffle spreads the sound in different directions.

Modifying electronic sounds with vibrato is one way of approaching the naturalness of some sounds, but it is not sufficient in imitating pipe organ choruses. Sustained organ tones interact in a complex way, producing a warm, indefinite movement called the choir effect. This result can be arrived at electronically by the introduction of noise to randomly control the tone oscillators. This is not the same thing as adding a little noise to the actual tone. The idea is rather to introduce random variation in the oscillators. A similar effect can be obtained by randomly affecting various vibrato oscillators, thus introducing different oscillation patterns among the many interacting tones.

Gross modifications of pitch and intensity of sound are easily obtained with electronic equipment. Wide variations of amplitude produce throbbing effects, and fluctuations in pitch create wa-wa and Hawiian guitar results. A continuous gliding of pitch, or portamento, can be produced directly on some electronic musical instruments such as the Theremin and Moog and portamento can also be produced by modifying discrete tone melodies by sending them through an integrator followed by a smoothing low pass filter.

Ring modulators and multipliers are popular devices for tone modification. The essential working of a ring modulator is the modulation of one signal by another. The ring modulator accepts two signal inputs and produces their sum and difference as outputs, which usually sound entirely different from both inputs. Multiplicative modulation of one signal by another can also be accomplished.

Very significant work in sound modification has been done with Max Mathew's MUSIC IV by J.C. Risset at the Bell Telephone Laboratories [123]. It is debatable whether this work should be classified under sound generation or sound modification because the computer generates the final samples directly. I have chosen to include Risset's catalogue of computer synthesized sounds in the present area of modification because the computer programs mainly manipulate arrays of numbers representing simple periodic waveforms and produce complex waveforms. All types of sound modifications can be done very precisely by digital waveform processing. Attack and decay envelopes may be defined, randomly varying or periodic vibrato and tremolo may be applied, additive synthesis or subtractive synthesis may be accomplished, all types of conceivable filters may be constructed, and thus unlimited timbres may be created. Risset has published a catalogue of computer synthesized sounds, some of which are new and bizarre. He has produced timbres that have the tone qualities of flutes, clarinets, brass instruments, pianos, and percussive instruments including bass drums, bongos, snare drums, and gongs. The simulation of a gong is especially interesting because random control has enabled the sound to have natural waverings as are heard from an actual gong due to the highly interacting waves in the metal that determine the different vibration modes all over the gong. New and unusual sounds created by Risset include those made by experimenting with various shapes of attack and decay curves, different envelopes for harmonic components of a sound and bi-directional glissandi. Two interesting features of this catalogue are that first it comes with records which contain examples of the sounds, and second the descriptions of the waveforms and modification parameters are described completely for each sound. This information is of course very valuable to anyone wishing to synthesize various sounds, whether electronically or digitally.

Sound Coordination and Organization.

The third major area of sound processing is the combining and mixing of sounds. This area is concerned with how various tones, timbres and melodies are organized and put together. There are three aspects of this area, which are rhythmic, synthetic, and spacial.

The rhythmic aspect deals with the organization of the pitch structure of a sound. This topic includes the assignment of durations to the individual tones. Usually this is done directly by the performer on an electronic musical instrument; as he plays melodies he automatically determines the rhythmic content. In the production of electronic music with tape recorders, the durations of notes are altered by different tape recorder speeds. There are recorders which will allow the speed-up of sound without the usual increase in frequency by rotating playback heads. The frequency of the playback is dependent upon the relative speeds of the tape and the head. As long as the same ratio is maintained for the playback head and tape speeds as was present between the record head and tape speeds at record time, the frequency will remain the same. Conversely, if the tape speed is not increased but the rotating head is made to rotate, the result will be an increase of pitch but not of duration. There are various devices which control the durations of notes and these instruments will be described under the control area. Suffice it to say, very complicated and sophisticated rhythmic patterns can be imparted to melodies.

The second aspect of the organization area is the synthesis of a polyphonic line of music from two or more melodies, which may be either monophonic or polyphonic. Again, this is done directly by a performer as he plays a polyphonic instrument. The usual procedure in electronic music studios is to use tape recorders to combine melodies. Especially if the composer has monophonic instruments to work with, he will mix many melodies together by recording and re-recording several times. An example of this technique was the preparation of Walter Carlos' selections on the record "Switched On Bach." The pieces were prepared using a Moog Synthesizer, which is essentially a monophonic instrument, as follows: first, great care and time was taken to set up the filters and envelope shapers so that a particular tone quality was produced (for instance, the sound of a harpsichord) then a two or four measure monophonic passage was played by Carlos on the keyboard. The notes were played an octave below the final pitch range and recorded at half the playback speed. Not only did the monophonic passages have to be contiguously connected, but other monophonic melodies had to be mixed to produce the polyphonic music. The effort took approximately a year with the crude manual tape splicing techniques.

The mixing of melodies is very easy if the music is stored in digital form, or if the sound-production controls are stored digitally. In both cases definite and exact starting places may be determined for the synchronization and mixing of melodies. For instance, melodies formed on the RCA Synthesizer are defined by digital control information punched on a wide paper tape. Not only can an exact starting place be found, but the speed of the paper tape can be varied and thus the tempo. The first model of the RCA Synthesizer was driven in synchronization with a disk recorder which was able to record on several tracks simultaneously. Melodies were mixed very simply by recording one on one track, another on a separate track, and playing them back simultaneously. The later RCA Synthesizer, Mark II, is now located at the Columbia University Electronic Music Studio, and it is synchronized with a multiple-channel Ampex tape recorder; in fact, the recorder was the first multiple channel tape recorder that Ampex made. Synchronization between the synthesizer and the recorder is done by two signals which are put on the tape. The recorder is started up a few inches before the synchronization clicks so that the tape speed is maximum when the first signal is sensed. This signal turns on the synthesizer tape which has been prepositioned to the desired starting place. The second click turns on the recording head on the specified channel to start the actual recording after the synthesizer tape has reached its normal speed. Two tape recorders can be synchronized by the same method, and this is much more precise and easier than the manual techniques for synchronizing tapes. While visiting the Columbia Electronic Music Studio I observed how Bülent Arel mixed electronic music passages from two tapes onto a third. He first listened to the two tape selections and decided how much of one he wanted to mix with the other. To determine where to cut the tapes he manually slid the critical portion back and forth under a playback head and listened carefully to find a good breaking point. After he had done this with both tapes he formed two loops with the sections of tape having the desired passages on them. Then he ran each loop through a different tape recorder and listened to the results. He varied the volumes of the individual recorders at the mixing panel then he varied the speeds of one then the other loop, always listening, I supposed, for interesting combinations of sound.. He had measured the two tape lengths and made sure they were not in a simple ratio of each other so that the sounds from each tape would interact differently as the loops continuously played. Finally he decided that a certain combination was worth recording so he started up a third recorder from the mixing panel and recorded the resultant mixing of the two tape loops for some time. I have cited this as an example of some of the classical studio techniques that are carried on in producing electronic music by mixing various recorded sounds. The tape loop routine is very commonplace and was probably introduced in America by Vladimir Ussachevsky, who was one of the pioneers with Otto Luening to bring electronic music to the United States [90]. Most electronic music has been produced by manual tape manipulations that include the splicing of different sounds together and the recording of one sound on top of another. The recording of one piece of recorded sound on another has been done in a wide variety of ways such as the variation of their relative speeds, the changing direction of one or both of the passages (playing the tapes backwards), the blending of one with the reverberated sound of the other, and the variation of their individual volumes [147].

The electronic music composer usually works entirely with tapes, synthesizers and other electronic devices to produce the music to be performed. Vladimir Ussachevsky introduced the combination of tape and live musician performances which were well received. The synthesis of the final production can be achieved by four means of supplying cues to the live performers: 1) each performer can play at random what he wants from the score and how fast he feels like playing it; 2) cues are supplied at periodic intervals; 3) special synchronization cues come to the conductor or performers who are wearing earphones; and 4) each player receives cues, but he decides how he will contribute to the musical effect [149].

The third aspect of the organization and combining of sounds is spacial. The difference between high fidelity and stereo illustrates an effect caused by the spacial mixing of sounds. The placement of sound sources in relation to the listener has a great deal to do with the character of sound perceived. The size, shape, and makeup of a room have much to do with the final sound effect from speaker outputs. The number of sound sources (speakers, for example) their frequency response, and the directions they face is very important, as can be seen with Bose speakers, which reproduce recorded sounds so well. These speakers are composed of several small loud speakers which are directed at different angles and are usually placed facing a wall so that the sounds from the speakers are further mixed and reflected against the hard surface. With this arrangement the entire sound fills the room and the "stereo" effect is heard around the room. In the ordinary stereo situation with only two speakers, there is a limited range where the stereo effect is prominent. Various productions of electronic music have utilized many speakers positioned all around the audience. Some speakers are placed in front, others on the two sides and behind, and still others above the listeners. Control panel settings determine which sounds are directed to each speaker system. The auditory output can be made to sound as though it were moving from behind the audience to the front or vice versa; it can sound as if it were rotating around the room one way and then the other, or it can appear to be coming from all over. An important idea is that the directions, flow, and sources of the sounds can be electronically controlled. Interesting effects are caused by sending one melody around the audience in one direction and another melodic line in the other direction, or in the same direction either lagging or leading, or traveling at a different speed. The spacial effects of sounds produce new mixtures and combinations which add variety and interest to electronic music [85].

Waveform Storage.

The fourth area of electronic music production is the storage of sound materials. The obvious and most widely used media are magnetic tape and record disks. Storage is important becuase it facilitates the saving of certain melodies and sounds, which may be later mixed with other sounds; and it makes it possible to serially process some sounds at different time periods. In a certain sense, all the tone generators are storage devices that yield up their sounds upon request. Paper tape, long sheets of punched paper, thin magnetic wires and computer memory have all been used to store music. Computer memory includes core memory, integrated circuits, rotating magnetic disks and drums, magnetic tapes, paper tape, and punched cards. Music is represented digitally on these mediums in four ways: analog waveform samples, analog control information, digital control data, and algorithms. Usually a combination of these ways is used. For example, in Max Mathew's MUSIC V and GROOVE programs waveform samples are generated by algorithms which store the information in tables, so a combination of the first and fourth storage methods is used. In a system at the University of Toronto,digital data representing analog control information is converted to analog voltages which control voltage-controlled filters and oscillators. Our music system at the University of Utah is an example of stored information used as digital control. The digital data stored in the computer is sent directly to a matrix of latches which control the switching of analog tone signals.

Core and integrated circuit memories are now being used by the Rodgers and Saville organ manufacturers to store stop-and-coupler settings. By this means a combination button can be programmed to determine a certain state of the organ stop settings and couplers. When the button is thereafter pressed, the appropriate state is invoked. If another organist wants the same button to specify a different state, he simply reprograms the button, putting the new state information in the memory. Subsequent depression of the combination button causes the stored information to set the new state of the organ.

Sound Control.

The fifth area of electronic music production is control. This area is most important because it directly determines the interaction of the four previous areas; namely, the generation, modification, mixing, and storage of sound. Control occurs in levels. For example, the hand can control capacitance which determines voltage levels, and the voltages can control oscillators, filters, amplifiers, and so forth. The different types of controls interact with each other in diverse ways. There are four types of control: manual, electronic, stored data and algorithmic.

Manual control consists of the manipulation of sounds by persons by means of keyboards, linear controllers, patch panels, mixer panel knobs, touch sensitive devices, velocity sensors, stop tablets and coupler switches. These devices are activated by movements of a person's hands, fingers, and feet. On the Theremin the positions of the two hands directly control the pitch and volume of the resulting sound. Modifications of this idea allow the position of the hand or finger on a wire or tube to control both pitch and volume. The pressure sensitive control tube on the later models of the Trautonium is an example of placement and force exerted by the hand in controlling the audio output.

The Moog equipment includes a linear controller which is a wire stretched over a plate of resistance. The position at which the wire touches the resistive plate determines a control voltage. The accompanying organ-like keyboard works on the same principle; that is, the depression of a key defines an output voltage. The circuitry is arranged in a serial fashion along the keyboard, so only one control voltage is determined even when a number of keys are simultaneously depressed.

Electronic organ keyboards, then, are used to control either monophonic or polyphonic instruments. There have been many keyboards designed for different purposes. For example, the leather bands with transverse welts on them were used as keyboards on various European single-melody instruments (Chapter 2). The leather bands were placed over stretched wires that could be pressed against resistance plates. The welts along the leather bands indicated the positions of a predetermined scale (the welts could be slid to different positions along the bands). Keyboards have been designed for different scales having more notes than twelve. Various scales have been proposed; for example, the Pythagorean scale was built upon perfect fifths, but it did not primarily contain octaves.

Name	C	D	E	F	G	A	B	C'	C"
Just Ratio to C	1	9/8	5/4	4/3	3/2	5/3	15/8	2	4
Temered Ratio to C	1	1.122	1.260	1.335	1.498	1.682	1.889	2	4
Pythagorean Ratio to C	1	9/8	81/64	4/3	3/2	27/16	243/128	2	2187/512

Figure 1-8
Scale Intervals

The last number in the Pythagorean row of Figure 1-8 is approximately 4.3, quite a bit different from an octave. The just scale is interesting in that it has two different whole tone intervals. When the ratios between consecutive notes are noted, we see that E-F and B-C are in the ratio 16/15; D-E and G-A have 10/9; and C-D, F-G, and A-B all have 9/8. The 16/15 ratio represents a semitone, and the 10/9 and 9/8 are the two whole tone intervals. These last two ratios differ by 81/80, which is called syntonic comma. If the difference of the comma is ignored, a just scale can be built up with 22 notes (2 different notes for $C^{\#}$ and $D^{\flat}$, etc.). An even better approximation can be accomplished with 50 steps per octave, and the next improvement comes with around 300 steps [30].

It turns out that just intonation cannot be maintained in all keys without unmanageable complexity. The introduction of sharps and flats makes things even more difficult because theoretically there are many kinds of F-sharps, etc.; and the F-sharps are not equal to the G-flats. Just intonation can be approximated fairly well with two equally tempered scales, one with A=440 and the other with A=436.3. The idea is to switch from one scale to the other depending on the key being played.

The tempered scale is constructed by dividing an octave into twelve equal parts and by spacing the notes such that each one is $\sqrt[12]{2}$ times the frequency of the preceding note. This construction, as can be seen from Figure 1-8, approximates the fourth and fifth intervals of 4/3 and 3/2 fairly closely. For instance, the perfect fifth ratio of 1.5:1 is approximated by 1.498:1. Intonation-wise, these are important intervals and they are approximated the same in all keys. Helmholz argued against equally tempered scales in his "Sensations of Tone" by saying that it would have a marked effect on our acuteness of appreciation for harmony. He remarked that the justly intoned chords "flow on with a full stress, calm and smooth, without beat." He continued, "Equally tempered chords sound beside them rough, dull, trembling, restless" [50]. Scales with 7, 12, 19, 22, 31, and 53 number of steps per octave have been suggested and some exotic keyboards have been built to control several of these scales of various lengths [30,126]. A novel design from the Netherlands was a thirty-one note organ keyboard which had terraced, interlaced keys colored white, black and blue. There were five divisions between each of the pairs C-D, D-E, F-G, G-A, and A-B, and three divisions between E-F and B-C. The individual notes were symbolized as follows: C, C+, $C^{\#}$, $D^{\flat}$, D^{-}, D, D+, $D^{\#}$, $E^{\flat}$, E^{-}, E, E+, F^{-}, F, F+ ... C^{-}, C. The 31 notes were chosen on the theoretical basis supplied by the famous Dutch scientist and musician, Christian Huygens [51]. The claim was that on the 31-note pipe organ one could play the music of every civilization.

The discussion of scales was mentioned briefly because modern electronic control makes it possible to use scales of various lengths and to investigate their characteristics. Paul S. Rosberger has been concerned for some time about the re-design of the common organ and piano keyboard. In 1966 he suggested a keyboard which required the depression of two adjacent keys simultaneously to produce a note [127]. Each key width was less than an average person's little finger, so the depression of a finger always played a note. The next year Rosberger presented an improved version of his keyboard idea [126]. In this model there are no actual keys to press, but strips of capacitive sensor elements which are spaced one centimeter apart. As Figure 1-9

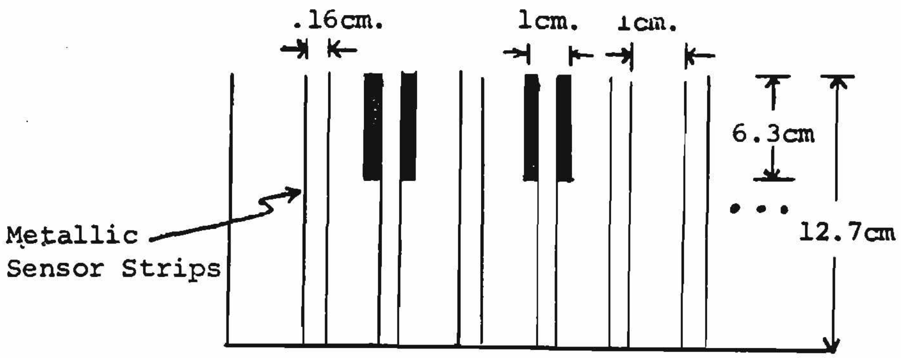


Figure 1-9

New Keyboard Design

shows, each note, natural or sharp, is represented by a similar strip. Those which correspond to normal black keys are notated by the black bars which are for identification purposes only. There are several advantages of this keyboard over the conventional one; for instance, there are no moving parts to break down, the keys are closer together so that a wider span of notes is possible to the player, chords in different keys all have the same hand position since whole tones and semitones are always a fixed distance apart and the keys can be made touch responsive.

Another new keyboard for controlling synthesized electronic music is the typewriter-like keyboard on the RCA Synthesizer [60]. The keys resemble the old fashioned typewriter keys which were visibly attached to long levers. The function of these keys is to punch tape-control information which directs the synthesizer as is explained below. In the situation with the paper tape preparation, the composer is not immediately controlling the music output, but he is storing the control information for later use. In the system at our university the teletype, or any other peripheral terminal connected to the computer, can be programmed so that it directly controls aspects of music output. For example, the teletype keys could be used to play a melody, alter stop settings or dynamically control volume levels, etc.

Other manual controls include patch and mixing panels with their accompanying switches, knobs, tablets, and sliding bars. In prominent electronic music studios where there is a lot of equipment it is important to have a mixing panel where the various devices can be connected together. I mentioned the mixing panel several times as I described Bülent Arel's composing techniques to illustrate this point. Patch chords, sliding bars, and switches can also be set up so that certain regulation voltages result. These will be dealt with under electronic control. Variable resistive keying, capacitive force-sensing switching, and velocity-sensing keying are all used to provide precise and responsive touch control to the performer.

Electronic control has become especially interesting and important since the development in 1964 by Robert A. Moog of voltage-controlled oscillators, filters, and amplifiers [110]. All of the Moog voltage-controlled modules are designed with similar input and output impedances, voltage and current levels and frequency response. Thus all the output signals from each module can be input into any other module. Control signals consisting of voltages determine oscillator frequencies, filter characteristics, waveform envelope shapes, rhythmic sequences and amplification levels. The control is linearized in most cases which means that an increase of one volt in the control signal produces a discrete output signal change which is always the same amount over the entire frequency range of the output signal [39]. Any electronic device which is able to produce an electrical signal of any type, can be used as a control device to produce and modify audio waveforms.

Stored-data control has become a significant means of organizing sounds and regulating equipment. The RCA Synthesizer can be controlled by punched paper which passes over a drum. Holes punched in columns along the paper specify the following sound parameters which determine the output sound: frequency (from 20 cps, to 20,000 cps.), intensity (0 - 120 db.), attack, duration, decay, timbre, portamento (free glide), frequency and amplitude modulation (vibrato for instance), and random effects [116]. Ordinary teletype paper-tape has been used to supply data to voltage-controlled equipment. The data is converted from digital form to voltage levels by a D/A converter, and the resulting analog signal operates the electronic devices. Any digital data, wherever stored, can be used in this way to control sound production. The natural consequence of this is that the computer will play an increasingly vital role in the synthesis and control of electronic music [66]. The processing power of computers will facilitate the creation of very complex mixtures of timbres, rhythms and melodic lines. Another tape driven device is the University of Toronto Hamograph [9] which is a programming device that has six control-information sensors. The control information takes the form of metallic foil strips which are attached to the surface of a tape film which can be driven at variable speeds. The control becomes digital as it is picked up by the sense wires, and the resulting signals are used to turn on various devices or to control sound modification equipment.

Stored data, then, can be used to synchronize electronic equipment. As was pointed out in the area on sound mixing, the synchronization of recorders, synthesizer drives, and other mechanical equipment is important. An idea from the University of Illinois is to record timing information alongside recorded audio, then synchronization can be done precisely by electronic controls [7]. The mixing of sounds from different tapes is done so much that it would be interesting to be able to have several tape controllers, all of which could be independently addressed and each of which could be synchronized by timing information stored along with the audio signal.

Digital control information is used in an interesting way by Max Mathews in his GROOVE computer program [95]. Manual control is sensed by the computer as digital control information which in turn modifies digital data that directly defines an output waveform. As the user enters control information, the computer processes the control, modifies the stored waveform samples, and sends out the resulting functions of time which may be used as analog control signals. The person composing the sounds is able to have sensory feedback in real time. The feedback response may be audio or visual (CRT display), or both. The composed functions of time may be used to control any electronic equipment, as described above. Stored data in our system at Utah can represent levels of control information. The bottom level controls which output switches are on (these affect pitch, volume, number of tones, and timbre) and other levels of stored data control this basic level. As mentioned before, stored data has been used in core and integrated circuit memories to control stop and coupler settings in a few electronic organs.

Algorithmic control is closely associated with stored-data control. In the case of algorithmic control, the data represents a procedure to be carried out. This procedure may be a condensed representation of stored data (a sine calculating routine as opposed to the waveform samples) or it may specify a complex transformation to be done according to certain conditions which exist. The programmer determines these conditions by inputting control information to direct the algorithms. In my system, as well as in the GROOVE system, the input data, whether from a keyboard, knobs, tablets, or light pen, may be interpreted in any way by stored computer programs.

Since control is such a major factor in all areas of sound synthesis, the computer assumes a vital role in electronic music production.

Music Notation.

The final area of electronic music organization is notation. Electronic music is a composer's world in which outsiders and newcomers feel ill at ease and lonely because there is a lack of a descriptive symbolic language. Attempts have been made and seminars held to [23] develop notations for expressing the procedures involved in creating a piece of electronic music. The description of how to perform certain electronic music is bad enough, with designated and random cues to live performers, direction to turn knobs at sundry times and specifications for spacial effects. These actions as well as tape manipulation procedures, electronic device settings, etc. are difficult, if not useless to notate. Vladimir Ussachevsky has patented an electronic music score writing notation that has certain symbols to specify filtering effects, noise introduction, envelope shapes, reverberation, etc. [147]. Junior high and high schools in Philadelphia encourage students to organize electronic music composition and then devise their own notations. Many of the students responded by writing poetry and creating accompanying art.

With the advent of computer-created music came further attempts at languages to describe music. At Princeton University, Regner and Robison developed SAM (System for Analysis of Music) and IML (Intermediate Music Language), respectively. Both systems are bent towards a linear representation of common musical notation. Information such as durations, pitches, slurs, ties and articulation marks are stored in arrays and accessed by special routines. The Ford-Columbia linear notation for ordinary music scores designed by Stephan Bauer-Mengelberg is probably the foremost development along these lines. The effort is intended for use with a computer-driven music printer, but the final form of the notation has not been fixed yet [65]. In this system as in SAM and IML, information describing various notational markings is extracted and put into tables which are indexed by time [47].

A comprehensive and successful program for defining sound structures is Max Mathew's MUSIC IV in which the notation specifies waveform parameters. The basic building blocks are routines which function as oscillators, filters, amplifiers, and mixers. Here the music representation consists of numbers which define various parameters for the routines that determine the waveforms [142].

At the University of Illinois in 1963 a computer program MUSICOMP was developed [5] that allowed the user to select various routines which yielded up random numbers from various sets and distributions. These numbers were used to determine the sound parameters, pitch, octave, loudness, duration, timbre, and type of attack.

On the RCA Synthesizer the music notation is expressed in tabular columns of binary encoded information [1].

Graphical plotting of sound parameters enables both the representation of continuously varying parameters, such as a glide in pitch, and of discrete events, such as notes and rests in a melody. Notation of musical output has been expressed in several graphical forms. For example, Chang and Mathews wrote a graphical score-writing program that plotted pitch and volume versus time of each separate voice sounding from the computer [21]. The ordinate values were scaled and normalized so that the graph of a voice having a wide range of pitches was not larger than that of a voice with a small pitch range. For example, in the former case the vertical distance of five inches on the graph paper may cover five octaves; whereas it could correspond to a single octave in the second voice. Ordinary music notation is mainly prescriptive in nature with an emphasis on the structures, pitch, and rhythms. Graphical notation on the other hand can be very descriptive [133] with precise information representing amplitudes, pitches, fluctuations in the basic pulse and attack envelopes.

Various new and specific notations have been developed and will continue to develop to represent express and describe sound production, modification, mixing, storage, and control.

CHAPTER 2

HISTORICAL DEVELOPMENT OF ELECTRONIC MUSICAL INSTRUMENTS

The development of electronic musical instruments can be broken down into four periods. The first includes the attempts at electronic music production before the discovery of the vacuum tube. The second period extends from 1920 to World War II and comprises the development of novel single-melody instruments such as the Theremin, Hellertion, and Trautonium, and the development of polyphonic instruments such as the Miessner electronic pianos and many electronic organs. The third period is an extension of the second period after World War II. This area covers the refinements and embellishments added to the instruments developed before the war. The basic ideas of nearly all the instruments we have today were developed in the period between 1920 and 1940. Since the war, the vacuum tube oscillators have been refined or replaced by transistor circuits; elaborate envelope shaping and tone forming circuits have been added, and novel percussion simulating circuits have been developed. The fourth period beginning in the 1950's includes the electronic music synthesizers which are controlled by means other than ordinary keyboard manipulations. These instruments include the RCA synthesizers and the Hamograph, the Moog, and the Buchla equipment.

Novel Experiments Before the Vacuum Tube.

The first serious attempt at the widespread production of music by electrical means was made by Dr. Thaddeus Cahill at the beginning of the 20th century. In 1903 he produced some music sounds with his giant Teleharmonium, or Dynamophone, which consisted of about 30 carloads of tone generators and controlling apparatus weighing around 200 tons. The vacuum tube amplifier had not been discovered yet and so large rotating magneto-electric generators were used to produce the alternating audio signals. The large dynamos, that measured about two feet in diameter and four feet high, produced near-sine waves which were selectively added by stop switches to synthesize the tones. Dr. Cahill had visions of transmitting his music to subscribers over telephone wires, but electrical engineering was not far enough advanced at that time to prevent inductive interference between the regular telephone lines and the Teleharmonium, so after many complaints from telephone subscribers, the costly project was abandoned [88,90].

There were early attempts before the vacuum tube was discovered to amplify piano tones by using telephone repeater amplifiers. In 1903 Farrington invented an instrument called the Choralcello which worked on the principle of additive synthesis. To produce various timbres, a number of harmonics were added to a fundamental tone. Even though some of the results were encouraging, the effort came to an end because of excessive cost and intricate and unstable equipment [40]. Duddell's "singing arc" demonstrated in 1899 was one of the earliest devices which produced electrical music [35]. The periodically discharging arc was part of a tuned circuit which could be made to oscillate at definite frequencies [34].

Early Developments Before World War II.

The development of the thermionic value, or vacuum tube, by Dr. Lee de Forest in 1908 opened up the door for extensive production of electronic music. In fact, de Forest himself made an electronic musical instrument in which a variable capacitance controlled an oscillating valve [35]. In America, Laurens Hammond and Benjamin F. Miessner showed the capabilities of the new amplifiers by using them in many patented musical instruments, including various "electric" pianos and organs.

In Europe, after the vacuum tube was discovered, it was used as both an oscillator and amplifier in many of their single-melody instruments. In these instruments one oscillator was controlled to produce all the output frequencies. For economical reasons, the attempts to produce instruments with large numbers of tubes failed. Descriptions of some of the instruments from the second period follow.

The Theremin was the early well publicized sound-effects instrument invented by the Russian Leon Theremin in 1919. It was capable of emitting horrifying wails and ear-piercing shrieks to smooth gliding tones. It worked on the heterodyning of two radio frequency oscillators which oscillate at different frequencies. There was a fixed oscillator and a variable one which was controlled by a varying capacitance. To begin with, the variable oscillator was adjusted to match the frequency of the fixed oscillator so that no beating occurred. The high frequency of these oscillators, of course, was out of the audio range. As capacitance was added to the tuned circuit which determined the oscillation frequency of the variable oscillator, the frequencies of the two oscillators interfered with each other, producing a frequency which was the difference of the two and which was in the audio range. The variable capacitance was added by the performer's hand which moved to and fro near an antenna. The other hand also introduced a variable capacitance which controlled another oscillator. This second oscillator was connected to the gain control of an amplifier and the closer the hand to the antenna, the less the output from the oscillator. This reduced the bias on the amplifier and thus increased the audio volume. In the space of about three feet the pitch range of six octaves could be covered. Tone circuits consisted of some isolating condensers and radio frequency transformers, which passed on or rejected certain harmonics produced by the oscillators. These high frequency harmonics produced corresponding overtones in the beat note, and the control of these harmonics facilitated the imitation of many tones of different musical instruments. One of the most realistic imitations was that of the cello [130, 14, 109, 106, 44].

Another instrument, the Trautonium, developed by Friedrich Trautwein in Berlin, Germany, produced periodic waveforms by means of oscillating neon tubes. These tubes were used to generate sawtooth waveforms, rich in harmonics. The frequency of the oscillator was determined by a tuned R-C circuit in which the resistance was variably controlled by the position at which a taut metal wire was pressed onto an underlying metal rail. The rail was a resistor, and the point at which the grounded metal string contacted the rail fixed the resistance value in the oscillating circuit. Dummy rubber keys were mounted over the wire to indicate the positions of the tempered scale. The keys could be moved to different positions along the wire to provide varied scales. The first version of this instrument was marketed by the Telefunken Co. in 1930. The early models contained a few formant circuits through which a limited assortment of tone qualities was produced. The use of formant circuits in this machine was an interesting application of the German developed formant theory. The formant circuits were essentially resonating circuits which introduced an emphasis to certain frequencies of the waveforms being modified. Various ordinary instrumental sounds could be approximated by the Trautonium's formant filtering [12, 79, 97].

An instrument resembling the Trautonium was developed by Helberger and Lertes called the Hellertion. It was a single tube instrument with a pitch range of five octaves. The oscillator frequency was controlled by a variable grid voltage which in turn was determined at the keyboard as in the Trautonium. The keyboard was responsive to touch, the finger depression determining the loudness of the tone. Touch responsive keyboards became common on many European instruments and after the war, elaborate mechanisms were created which controlled the attack rate and volume of the sounds [88,41]. One such control is described later in connection with Sala's Mixturtrautonium.

One of the troubles with the Theremin was that a new technique, hand movement, had to be employed to "play" the instrument. The Trautonium took a step in the direction of keyboards with its dummy keys, and the Hellertion followed suit by having transverse welts in intervals along long thin strips of leather. The welts indicated the position of the equal tempered notes. A wire running under the leather strips corresponded to the taut wire of the Trautonium. Another problem with the Theremin was that it could be played only in a glissando mode, where all intermediate pitches would be sounded as the performer moved his hand from one note to another. Jorg Mager eliminated this restriction in his Sphaerophon which like the Theremin, used the beating of two high-frequency oscillators to produce the audio tones. Mager's idea was to switch various amounts of capacitance across the circuit controlling the variable r.f. oscillator. The tuning capacitors were arranged in parallel with a key-operated switch between each section.

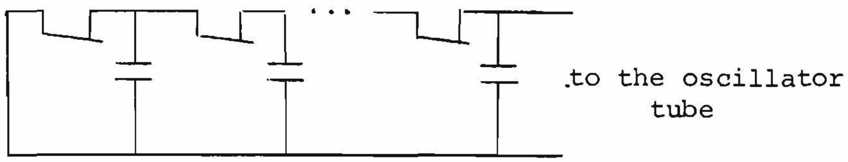


Figure 2-1
Variable Capacitance Switching in the Sphaerophon

Depression of the key opened the switch, thus varying the amount of capacitance in the circuit. The Sphaerophon was first demonstrated in the Donaueshingen Festival in 1926. There were four manuals (strips of leather) each of which was a monophonic "keyboard", so the performer could play up to 4 tones simultaneously. The Timbre qualities rather than being formed by electronic audio filters were produced by a variety of loudspeakers with differently shaped diaphragms, horns and other resonators. These resonators in turn added formants to form different Timbres [90,79].

A further instrument operating on the heterodyning principle was the French Le Ondes Martenot built by Maurice Martenot. In this instrument the capacitance was varied by the movement of a finger ring over the keyboard. There were also some stop keys which selected various filters. In 1928 the Ondes Martenot appeared in the Paris Opera, and ten years later a special version was built to produce microtones required in some of Hindemith's music [78].

The early European electronic instruments were mostly monophonic because the people were reluctant to build with a large number of tubes. The French Givelet-Coupleaux organ was an exception however.

The large French Givelet-Coupleaux Organ, pioneered in 1930, was the first experiment with a large number of oscillators. It had hundreds of keyed oscillators and its design set a precedent followed by the Allen and Conn Organ manufacturers, that is, a single oscillator per tone. The complex waveforms produced by the oscillators were <u>first</u> amplified and <u>then</u> shaped by filters and formant circuits. The idea was that since more current became available after amplification, the tone shaping circuits could perform greater phase shifts and thus more easily color the sound. The organ turned out to be bulky and it consumed much power. It was so expensive that the European market couldn't afford this design [12,40].

Captain Richard H. Ranger displayed his magneto-electronic organ in 1931. His system, which worked on the same principle as Cahill's Teleharmonium was much smaller due to vacuum tube amplification of the sound. Additive synthesis determined the tone color, and certain of the complex sound components could be modulated in amplitude. Decay shaping of the waveform envelopes produced percussive-like tones. There were twelve separate sets of motor driven alternators whose speeds were controlled by tuning fork vibrations [102].

Rotating discs provided a basis for periodicity, and before World War II, experiments were performed to produce tonal waveforms with gramophone records. Equispaced tinfoil waveforms were attached to the records and fixed electrostatic pickup electrodes were placed above the rotating waveforms. Later, an all metal disc was substituted with electrostatic pickup, but these experiments never proved to be fruitful because of mechanical instabilities which caused modulation and distortion of the tones [81].

In the late 1920's Langer produced his Emicon which used variable resistance selected by a keyboard to control the oscillating frequency of a neon tube. Later in 1934 Vierling and Kock produced their "Electroakustiche Orgel" which generated complex waves from neon tube oscillators. The complex waveforms were filtered by an inductive discharge circuit capable of producing organ sounds ranging from flutes to strings [102].

Around the 1830's the Frenchman, Constant Martin, developed the Clavioline which was a melody instrument containing a univibrator tone generator and two frequency dividers of the Eccles-Jordan type. Vibrato was controlled by a multivibrator which had variable oscillation rates. This instrument could imitate many orchestra sounds with its tone shaping and percussive circuits. Its appearance resembled the Hammond Solovox although the circuitry of the Clavioline was much simpler. It had a three octave keyboard and a five octave pitch range, so only a portion of the instrument's range could be played at one time [40].

The Neo Bechstein piano was a miniature grand having relatively short strings which were struck by very small "micro hammers." There was no sounding board, but instead the string vibrations were picked up electromagnetically and electronically amplified. The loud sounds became quite blurry, however, because of interference caused by spurious magnetic fields. The pickup device was connected directly to the vibrator and dampers could control the loudness of pitch [77,102].

Benjamin F. Miessner experimented with many ways of using electronic devices to produce music. There are several patents under his name displaying the sundry designs of electro-mechanical pianos, photo-electric organs, and other electronic musical instruments. Miessner developed an amplified piano which worked on the principle of the condenser microphone. A large voltage (150 v.) was impressed across the piano string and a pickup screw. As the string vibrated, the changing capacitance varied a voltage which was fed to an amplifier. The volume of the sound was determined by the distance of the pickup screw from the string. Each pickup was adjusted to give an overall balance of the 88 piano tones [103].

Another similar instrument was the Dynatone which had a knee lever that allowed the player to switch from harpsichord to piano to concert grandpiano sounds. The effects were caused by the degree of amplification and the number of harmonics added to produce the sound. With no amplification the struck notes resembled a harpsichord since there was no large sounding board. It was advertised as "an electric piano with radio and phonograph reproduction [82]. Along with the piano the console contained a record player, a radio, a microphone and speaker system.

The Electone was similar to the Dynatone except that the Electone had three electrostatic pickup electrodes along each of the 88 strings instead of just one pickup. Since the pickup screws were at different positions along the strings, several modes of the string vibration were sensed. Each of the resulting waveforms contained different harmonics and various tone colors were synthesized by the selective mixing of the waveforms. In other similar instruments selective harmonic content was achieved through the use of variable shaped pickup plates. Large pickup electrodes were used to filter out some of the high harmonics and thus enhance the lower notes. The first electronic pianos were introduced before World War II, but with the bombing of Pearl Harbor came the cessation of manufacturing. After the war, better models were produced [97,140].

Miessner also produced several instruments having electrostatic pickup of wind blown reeds. By controlling the polarities and magnitudes of the currents in the pickup electrodes, distorted, complex waveforms were obtained from the vibrating reeds. Thus variable timbres could be produced by voltage regulating potentiometers. The reeds also offered a delayed attack since it takes time to get them vibrating. Miessner was very happy with these instruments because the reeds were cheap, easy to use and could be keyed by simply opening air valves. There were also no problems of exact speed control or complicated timing mechanisms. Miessner was constantly seeking after the ideal instrument which was for him one that "can make any sound, known, unknown, or conceivable" [102].

The Loar Vivitone also used reeds as tone sources, but the vibrations were picked up magnetically as in the RCA electric carillon and in various stringed instruments.

In Europe Vierling produced the Electrochord, an electronic piano which had electrostatic pickups similar to the Miessner pianos. On each string several pickup electrodes were used and they differed in size. Some of them were placed under the bridge to pick up the attack transients. The formant tone control filters and envelope shaping circuits enabled the Electrochord to produce a variety of tone colors from the sound of a xylophone to that of an organ [97].

Just as the Everett Pianotron produced music by electrostatic screw pickup of vibrating strings, the Orgatron generated sounds from the electrostatic pickup of wind blown harmonium reeds. Screw pickups were adjusted over the five ranks of ordinarily shaped organ reeds to develop the proper volume. Each timbre had an independent bank of reeds, which were coupled pneumatically. Differently shaped pickup screws were experimented with and it was found that a flat head screw produced the most capacitance. To produce complex waveforms the languids of the reeds were shaped as was common to reed organ builders. Figure 3 shows some examples of shaped reeds which vibrate in different modes [101].

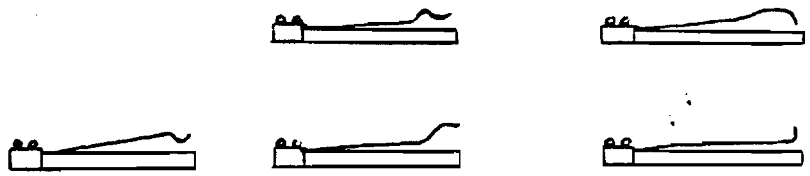


Figure 2-2
Shaped Reeds

Harold Bode, now living in New York, is the inventor of many electronic music instruments. He was especially active in building electronic instruments in Germany just before and after World War II. Since the European economy couldn't support large expensive instruments, inventors had to make several compromises in building consumer products. One interesting design was attempted by Bode in his Warbo Formant Organ. The objective was to have a multitone organ with few oscillators. This was accomplished by sharing the four relaxation oscillators such that four notes would sound simultaneously. The switching system assigned the first oscillator to the highest note, the second oscillator to the next highest and so on. The manual had 44 keys and there were two sets of filters and some envelope control circuits for percussive sounds. The venture was unsuccessful because the switching tree was too expensive and the oscillators weren't very stable [12]. In the late 1930's, Bode developed his Melochord which contained complex tone and envelope shaping circuits. An interesting idea employed in this instrument was a traveling formant circuit which was retuned as notes from different pitch ranges were played. The result was that a similar timbre could be maintained over the entire pitch range of the instrument. As an analytical device the Melochord permitted the investigation of different rates of attack and decay, imitative tonal synthesis, artificial echo and reverberation. There were two manuals of which the lower supplied the pitch note and the upper provided any other note with or without harmonics [29].

In 1932 the Russian physicist Ivan Eremeef demonstrated an organ which derived its tones from a system of electro-magnets which sensed circle. Some discs contained sine waves instead of recorded complex waves, and additive synthesis was accomplished to produce various tone qualities [40].

As early as 1921 an idea came from France to rotate the scan slits instead of the waveforms. In this design Toulon used stationary complex and simple waveforms. There were four light sources, each of which supplied two wheels. Shutters operating at the focal points of the light beams were controlled by the organ keys. The mask disc consisted of several sectors, each of which contain a specific waveform. There were twelve concentric circles of these waveforms so that an octave of tones appeared in each sector [101]. (See Figure 2-3).

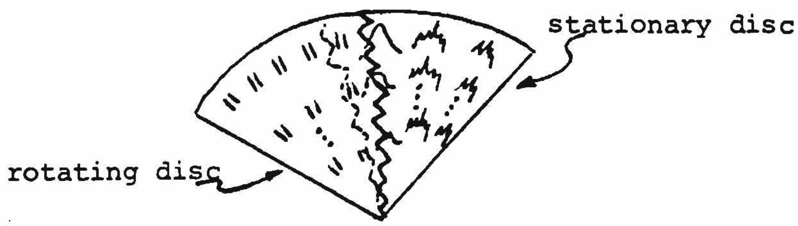


Figure 2-3
Toulon Disks

The other early full scale photoelectric organ besides the Toulon was the German Welte which was produced in 1930. In this design, the slits were stationary and the glass disc containing engraved waveforms was rotated. The waveforms were complete tonal images made from sound tracks of actual organ pipes. There were twelve identical tone wheels which were rotated at different speeds to maintain a tempered scale. Keys operated magnetic shutters which selected waveforms from one of magnetic fields set up by rotating toothed magneto-type phonic wheels. Earlier he had unsuccessfully experimented with his Photona, a photoelectric organ using perforated discs driven by a synchronous motor [40]. There have been many attempts to use the photoelectric effect to produce music. The ideas centered around a rotating wheel which contained evenly spaced holes, periodic waveforms, or recorded sound tracks.

One of the earliest patented photoelectric organ designs was by B.J. Miessner in 1926, after four years of experimentation. This system consisted of a rotational scanning mechanism which could generate sinusoidal and complex waves. The intensity of the light beam was varied to control the volume. This project didn't come much farther than the design state, probably because of inherent difficulties encountered in the photoelectric approach and because Miessner was having more success with other forms of electronic music production [102].

Another early attempt at using the photoelectric effect was made in the Hardy-Goldthwaite organ which used photographic discs containing variable density waves. The waves were replicated from those made by original instruments, and each disc contained the pitch frequencies of 71 notes of the tempered scale for a particular tone color. The rotating waveforms were scanned through optical slits which were controlled by shutters. Since the waveforms didn't quite match up at the ends when laid out in a circle on the disc, there was a click corresponding to the rotational speed of the disc when the shifting was done only at one place. In order to minimize this clicking effect, the shifting was divided up and done in several equispaced intervals around the

the 18 concentric circles of engraved images. The organ was received in Berlin with much enthusiasm. Unfortunately, the Welte organ was destroyed in World War II [25,32].

The Toulon idea of rotating slits was also used by Lesti and Sammis in their Polytone. The slit spacings were equal to the lengths of the waveforms, and thus the waveforms were repeated at the frequency determined by the number of slits passing per second. The waveforms were stored in radial sectors on a disc, and waveshapes were selected by rotating the appropriate sector of the disc to an operating position. Several rows of lights facilitated the retrieving of sounds from several sectors simultaneously [40].

Although many attempts have been made at using photoelectric generators, there were inherent difficulties that prevented this method from becoming widely used: Early efforts used 60 cycle alternating current to run their lamps because of the high cost to rectify enough current to operate the many lights. Unfortunately, this alternating current introduced a 120 cps frequency which modulated the synthesized sound. The emulsions containing the waveforms on the discs would contract after some time, leaving a distorted waveform. Lamps burned out and had to be replaced frequently. Finally, the equitempered notes of the scale could merely be roughly approximated, and then with tedious effort since the only exact frequencies possible with rotating discs are frequencies at integral multiples of the rotational speeds.

The only organ developed before World War II that still is being widely sold today is the Hammond Organ developed by Laurens Hammond of Chicago in the 1930's. He was a very successful inventor and holder of many patents even before he turned his talents towards the organ industry. He developed a synchronous motor which he used to drive clocks. One day, it is reported, he heard a humming noise coming from one of his clocks and decided he would build an organ using his synchronous motor. The tone generators consisted of 91 scalloped discs which rotated near the end of a permanent magnet with a coil wrapped around it. As the high spots on the edge of a disc passed the magnet, its magnetic field was varied which induced a small current in the coil. The tone wheel driving mechanism was so well constructed that present day models using tone wheels have the same design with little modification. The whirling tone wheels generated near-sine waves which were mixed to produce sounds by additive synthesis. The distinctive Hammond sound comes from the tones being generated all in phase and from the mixing of tempered harmonics rather than pure harmonics. Tone colors for each manual were controlled by a separate set of nine drawbars which determined the amplitudes of the fundamental and eight harmonics: the subfundamental, subthird, fundamental, second, third, fourth, fifth, sixth, and eighth. Every one of the musical frequencies in the entire range of the organ was produced by a distinct tone wheel. Each gear on the driveshaft was coupled to two tone wheels [32,43]. (See Figure 2-4).

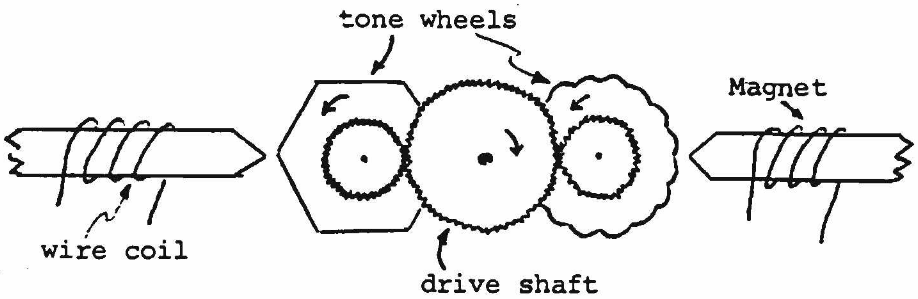


Figure 2-4

Hammond Tone Wheels

The signals from the generators were mixed, amplified, and then selectively sent to vibrato circuits.

On the very early Hammond organs the vibrato was achieved by amplitude modulation (tremulant). Later an interesting phase shift circuit was developed which imparted both frequency and amplitude modulation to the signal. A capacity pickup scanner tapped off signals from different points along an electrical time delay circuit consisting of a number of low pass filter sections. At each tap on the line the phase of the signal is retarded in relation to the previous tap. The scanner sequenced repeatedly up and down the line causing an undulating modulation of frequency. The amount of vibrato could be varied from narrow to wide by causing the sequencer to scan small to large portions of the delay line in a given amount of time.

The development of a reverberation control for the Hammond was also interesting. This device produced damped echos by reflections of waves within a network containing five lengths of springs, four tubes filled with oil, and two rocker arms. The electrical signal was transduced to a mechanical vibration which was propagated and reflected in the series of spring coils, some of which were immersed in oil to effect damping. The output vibration was converted by Rochelle piezo crystals to an electrical signal which was mixed in differing proportions with the original unreverberated signal [40, 104].

Laurens Hammond produced many other interesting designs in the organ industry. One successful invention in 1937 was the Novachord which used something like 163 vacuum tubes to produce the tones. It was the first commerical purely electronic musical instrument possessing a full keyboard on which chords could be played. Twelve vacuum tube oscillators fed the signals to frequency dividing circuits which produced outputs having a rich content of harmonics. These complex wave signals were then sent through envelope-shaping and tone-forming circuits which were controlled by dials above the single six-octave keyboard. The tone control dials had settings which allowed the player to select gradations of attack and decay from slow to fast, differing resonating filters to produce formants, various high and low pass filters, four levels of volume, and varying rates of vibrato. As a result, tones ranging from those of a plucked string to those of a bowed string could be selected, and the tone could be "full", "brilliant", or "deep" [77, 98]. George Gershwin was so impressed with the prospect of this instrument that he bought the first one and, of course, gave the Novachord much publicity [114].

Another Hammond instrument having tone controls similar to the Novachord was the Solovox which was a monophonic instrument. It was designed to be attached to a piano to provide an instrumental voice along with piano accompaniment. Attack and decay rates were determined by different sized capacitors which were switched in and out of the circuits by tone control knobs. The five octaves of pitches were generated by a single tube oscillator and four stages of frequency dividers. A panel switch was used to select which pitch range (bass, tenor, contralto, soprano) should sound. Any or all of these octaves could be used. Filter circuits enabled the Solovox to imitate the sounds of stringed and wind instruments including violins, cellos, trumpets, flutes and tubas [32, 99, 137].

The principle of magnetic induction was used in other instruments besides those of Hammond. John Goodell and Ellsworth Swedien developed their Mastersonic which used various shaped magnet ends to provide distinctive tone colors. There were twelve shafts with seven pitch wheels each which rotated near the irregularly shaped magnets wound with coils. Each of the pitch wheels contained twice as many rectangular teeth as the preceding one, so seven octaves were produced per shaft. Several differently shaped poles were dispersed radially around each wheel.

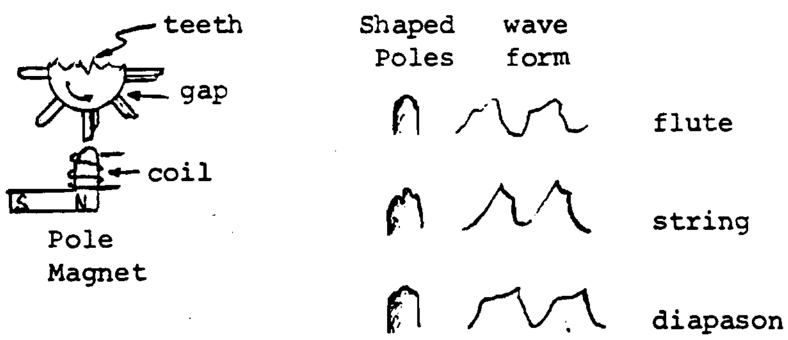


Figure 2-5

Mastersonic Tone Generation

As is seen from Figure 2-5, the different organ sounds were formed directly rather than by additive synthesis as on the Hammond. The aim of this Mastersonic project was to produce pipe organ sounds, and much of the design time was spent determining the shapes of the mechanism that simulated attack effects. The attack transients were obtained by extra contactors during the initial travel of the key. The resulting tone qualities imitated pipe organ sounds very well [62].

In England, John Compton developed a successful electronic organ which produced sounds by rotating electrostatic generators. The generators consisted of two plates, one stationary and the other rotating. The rotating member resembled to some extent a spider's web, consisting of a rim with radial spokes. These conductive radial lines rotated opposite the fixed stator which was a disc containing 5 concentric circles of engraven metal in the shapes of different complex waves. The inner annular ring contained two such waves and each succeeding ring contained twice as many as its predecessor. The waveforms were electrostatically picked off the stator by the rotating disc and fed to the grid of an amplifier. From there the signal was sent through the voicing filters. Vibrato was applied by slightly oscillating the stators [32].

Another organ, closely resembling the Compton but much smaller, was the Melotone. The tone synthesis was similar to that done on the Hammond since only sine waves were generated. Vibrato was applied by moving an eccentric pulley back and forth. The pulley was attached to the belt which drove the rotors [25].

Extensions of the Early Developments.

After World War II, improvements were made on the basic instrumental designs just mentioned, and these modifications make up the third historical period of electronic musical instruments. Oscar Sala extended the Trautonium to a studio model, the Mixturtrautonium, which had two sets of varied tone circuits, several frequency dividers; a foot-operated mechanism to provide gradual timbre changes and a pressure sensitive keyboard which controlled volume as well as pitch [74]. Sala was a student of Trautwein and Hindemith at the Berlin Hochschule für Müsik in the mid-1950's. The touch responsive keyboard had a liquid resistor in the form of a long tube which was depressed by the keys to generate gradations of volume and the rate of attack [33]. The liquid resistance tube was inside a long cylindrical grid resistor which controlled the oscillation frequency of the sawtooth producing thyratron tube. Around this grid resistor was a springy metal gauze which made contact with the grid resistor when depressed by a dummy rubber key [31].

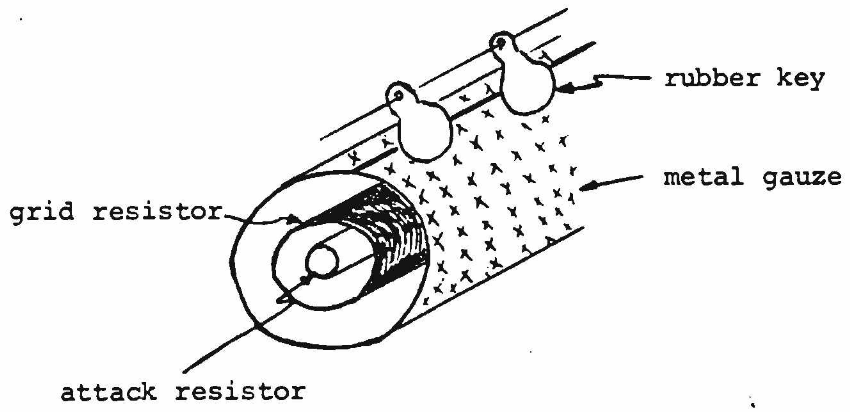


Figure 2-6

Thus an undulating movement of the fingers on the key would produce a vibrato consisting of both pitch and volume variation. The harmonic content of each octave was sent to independent tone filters, which meant that the lower octaves could be reproduced along with the fundamental pitch but with different tone coloring. The studio Trautonium was used to create the music of the ballet "Electronics" which made its debut in 1960 with 31 loudspeakers. The frequency of the performance ranged from 10 cps. to 40,000 cps. and the peak output approached the threshold of pain. The music of the ballet was described by critics as being conservative symphonic music. The Trautonium is capable, as a demonstration record shows, of producing the sounds of a mine explosion, steam locomotive, small town band (amusingly out of tune) and a full military band [85].

After the war Harold Bode produced a variety of interesting electronic music instruments including a large organ which used Hartly vacuum tube oscillators in conjunction with frequency dividers [12]. In 1953 he developed a small portable purely electronic organ called the Tuttivox which used phase shift frequency dividers. The Tittivox was later combined with a Clavioline under the name of Combichord and is manufactured by the German Hohner Corporation. Matthew Hohner, the founder of this company, specialized in making harmonicas and later produced several electronic novelties. One of these was the Multimonika which consisted of 2 manuals each with 41 keys. They also make the following electronic music instruments: the Electronium, a single melody instrument; the Hohnerola, a single manual reed electronic organ; the Cembalet, a harpsichord-like instrument with percussive tones; and the Hohner Vox and Bassophon, which are supplements for the Hohner accordians. The Bassophon, which had controllable attack circuitry, could simulate either a string bass viola or tuba in a 2-1/2 octave range. These instruments contain various formant filters which emphasize certain frequency bands and change the timbre of the accordian sound. In the Electronium the bellows serve a double function. Pressure at one end controlled the volume, while pressure at the other modified the harmonic content and thus the timbre of the sound [29].

Another instrument from Bode, the Univox, was an interesting single-melody instrument that produced sawtooth waveforms by the rapid charging and the slow discharging of various-sized capacitors which could be switched in and out of the oscillating circuit. Vibrato was furnished by a multivibrator which could be switched from four to eight cps. The vibrato oscillator could also be switched to trigger repeat-percussion to give effects similar to a mandolin, etc. As on the Clavioline the Univox had a three-octave keyboard and a five-octave pitch range. The keys were touch respondent in that a soft depression would yield a gradual attack and a quick, hard depression would cause a staccato effect. Percussive delay circuits and formant circuits enabled the production of various timbres [40].

Later a Univox-like instrument was added to an accordian to form the Hohner Multimonika. The upper manual controlled the monophonic melody while the lower operated the accordina accompaniment which was produced by 3 sets of wind-blown reeds.

One of the last electronic reed organs was the Wurlitzer which had several capacitive screw pickups above a wind blown reed as shown in Figure 2-6.

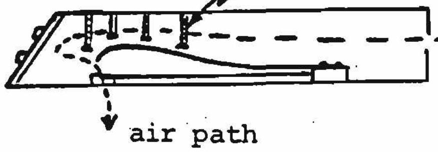


Figure 2-6

Wind Blown Reeds with Electrostatic Pickup

The air flow had to be carefully controlled because an explosive surge would cause the reed to strike the pickup screws [32, 40]. After this model, Wurlitzer joined the electronic group and began producing purely electronic organs. They do, however, still make electric pianos which operate on capacitive pickup and amplification of the vibrations of metallic reeds which are struck with key activated mallets.

The other means of electronic music production that held on into the late 1950's was photoelectric generation of tones. Many experimental designs with rotating disks were proposed. Figure 2-7 shows one such design. Five identical tone wheels are rotated, each twice as fast as its predecessor to give five octaves of tones. There are twelve concentric circles whose spaced holes determine the frequencies of a 12-tone scale. The wheel shown approximates an even tempered scale. Various lamps are turned on behind the wheel and the pulsating light beams are converted into oscillating voltages by the recipient photocells. Volume is controlled by interposing a diffuser between the light source and the photocell. The diffuser is a graduated exposed

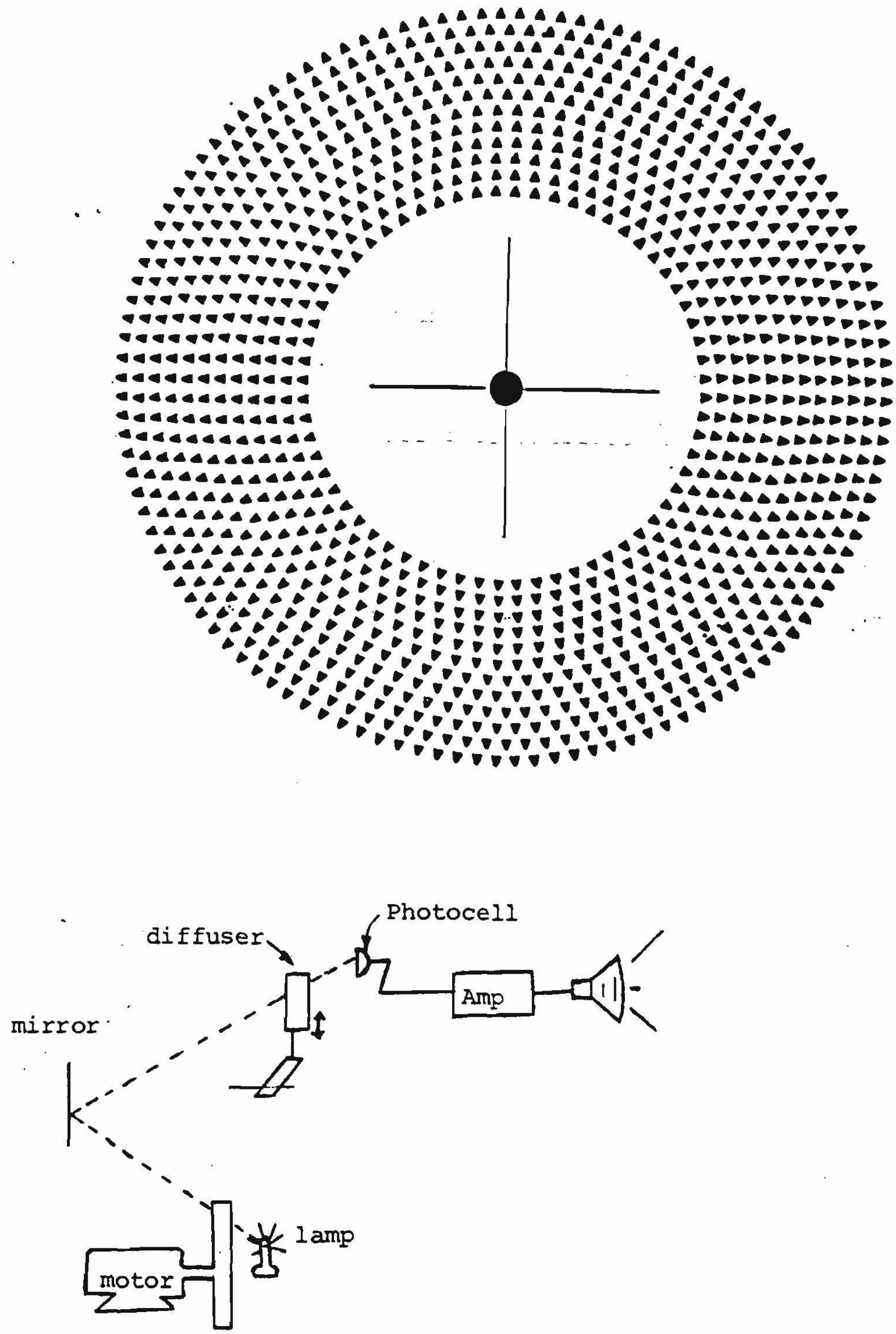


Figure 2-7

Photoelectric Disk and Tone Production

film which lets varying amounts of light through. A slight periodic movement of the diffuser produces tremolo [19]. Another photoelectric configuration utilized tone wheels which contained the fundamental frequency and nine other harmonics in successive concentric rings. Additive synthesis of these sine waveforms was accomplished by selective turning various lights on and off. A variable speed motor was used, but the instabilities in the rotational speed caused considerable audio distortion [64].

Baldwin produced a photoelectric organ in the 1950's, but quickly withdrew it. The last serious attempt at photoelectric reproduction of music was by the Kimball Organ Company in 1958. In this design a key selected the appropriate tone picture to be scanned by rotating slits. Each picture plate contained 61 recorded copies of actual pipe organ sounds [63].

In California, Chamberlain proposed an organ made up of a bank of tape recorders which played pre-recorded organ sounds, bass accompaniment, and special effects. The Mellotron, as it was called, had seventy individual 3-track tapes, each forty-eight feet in length. The keyboard was arranged in two 35-key manuals, side by side. The left 35 keys selected basic chord sequences and rhythmic patterns and the right keyboard played the lead instrument. The choice of 18 instruments included clarinet, flute, trombone, piano and organ [119].

The mechanical rotating and vibrating electronic musical instruments are a thing of the past now, with the exception of the Hammond rotating wheels. All large manufacturers, including Hammond, have turned to pure electronic sound production because of inexpensive solid state components.

Some very interesting things have been done in the electronic organ industry to make their instruments appealing to the consumer. An early development after the war was Hammond's chord organ which had a matrix of 96 buttons which could sound the 6th, 9th major, minor, 7th, diminished, augmented, and minor 7th chords of all the twelve keys. Each adjacent two or three notes shared the same oscillator, so the 32-note compass of the organ was covered by only 16 oscillators [26]. The first organ produced by Thomas in the 50's shared one oscillator for three notes. The sharing of oscillators in this manner is long outdated now.

The normal square, sine, and sawtooth waveforms were produced by a variety of ingenious techniques. Baldwin came up with the idea of staircasing square waves to approximate sawtooths and outphasing sawtooths to produce square waves. Staircasing is performed by mixing together square waves of two or more frequencies which are octavely related, synchronized and have amplitudes inversely proportional to frequency.

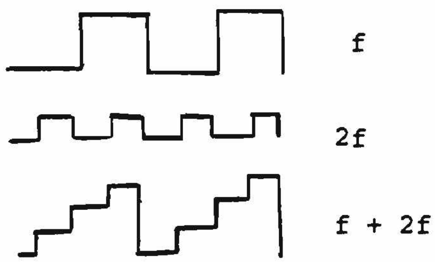


Figure 2-8
Staircasing

Figure 2-8 shows the simple staircasing of a fundamental tone and its first octave. The next note which could be staircased is 4f at one-fourth the amplitude of f. As more octaves are added in this manner, the resulting waveform contains a greater number of even harmonics. To see how the even harmonics are introduced consider the addition of f and 2f square waves: the fundamental contains harmonics f, 3f, 5f, 7f ... and the octave higher note contains 2f, 6f, 10f, 14f ...

Outphasing consists of adding one-half the amplitude of an inverted sawtooth at twice the frequency to the fundamental sawtooth. The result is a square wave. The interest of producing sawtooth waves is that they are used for producing diapasons and strings. Square waves on the other hand,contain only odd harmonics and are used for hollow woody sounds like that of a clarinet [25,27]. Square waves are directly produced by flip flop divider stages, and staircasing is an easy way to form a more useful waveform. Conn organs derive three

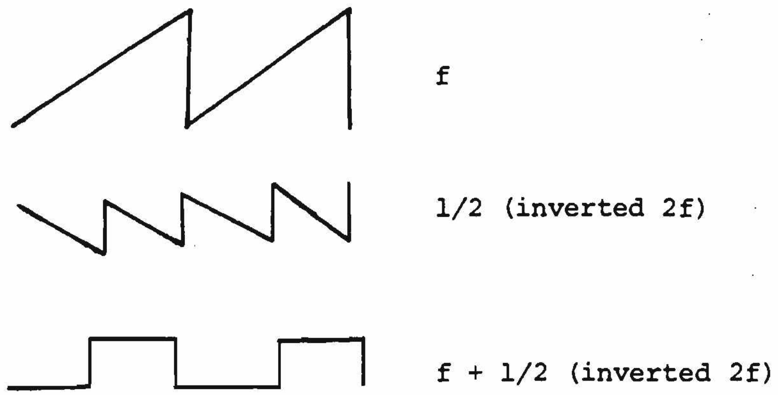


Figure 2-9
Outphasing

waveforms directly from their oscillator circuits. Both square and sine waves are keyed by transistors and the sawtooth is keyed directly. Conn used to have individual oscillators for each pitch, and with eighty-five oscillators they were able to produce ninety-five pitches.

The trick was to tap an extra octave note from the twelve highest oscillators by making a full wave rectifier which oscillated at twice the frequency [25]. Both the Allen and Rodgers organs have individual oscillator banks for each voice, but the unit principle is used. This means, for instance, that the 16', 8' and 4', etc. flutes come from the same rank of oscillators, each of which produces only one tone. The drawback to sharing a unit rank is the following.

Suppose the 8' and 4' stops of a rank are on and the keys C_4 and C_5 are played. The 8' stop causes the C_4 and C_5 tones to sound and the 4' stop causes only C_6 to be added and not another C_5 because C_5 is shared and already sounding. Saville organs get around this problem by supplying a rank of oscillators for each pitch register and each oscillator can drive several voicing circuits. In the Allen and Rodger organs, the oscillators are keyed, and in the Saville, audio signals are keyed. In each of these organs the voicing circuits are active formant filters which give naturalness to the sound. Chiff and wind noise is added to simulate pipe organ sounds. Chiff is accomplished by adding some of the 12th harmonic to the fundamental at the beginning of the note. The oscillators (Allen and Rodgers) or the active filters (Saville) are keyed slowly so that the gradual attack of an organ pipe is heard. Rodgers adds noise directly into the audio signal to give randomness and Allen modulates the oscillator keying voltage with noise, so the signal is wavered. This Allen Electron Whind is very effective in producing the kinds of fluctuations which are heard from pipe organs caused by varying pressures and random movements of air through the chambers [25,62]. Less expensive organs generally have twelve oscillators and generate the rest of the tones by frequency division.

There has been special emphasis among most organ producers to give the public a lot of bells, whistles, and gimmicks. Some of these are very interesting from an electronic simulation standpoint. Old theatre organs used to have a band box with percussion instruments inside which were activated pneumatically from the organ console. The methods have changed and the band box is still there, but highly sophisticated. There was a gradual development from the pneumatically operated percussion instruments to the modern completely solid state version. The Artisan Band Box was modeled after the older version in that genuine percussion instruments were used. Each instrument was operated by a solenoid, and the sounds were picked up on a condenser microphone and amplified [154]. The instrumental sounds included a cymbal crash, drum roll, cymbol tap, wood block, triangle, and cowbell. The Schulmerick ChimeAtron was also mechanically operated. The faint sound of vibrating bars which were suspended at the ends by nylon chords was picked up through inductive coupling and amplified.

Wurlitzer, in the meantime, put their brilliant electronic engineers to work to come up with some electronic voicing circuits which could imitate percussive sounds [75]. The result was the Wurlitzer Side Man which used the filtering of low audio signals (drums) and noise (cymbals) to produce impressive sounding percussion sounds. The percussive voices could be organized into rhythm patterns which included latin beats, waltzes, the fox trot and a march. The percussion could also be in a Ssh-Boom mode where drum beats and cymbol crashes sounded with pedal notes and cymbal sizzles played with the manual notes.

The Seeburg Organ Corporation after purchasing Gulbransen and retaining their name in 1966, produced the Select-a-Rhythm, which had the capabilities of 18 different rhythmic patterns and a selection of ten different percussive sounds. Earlier programmed rhythms (Rhythm King by Kinsman and Side Man) were sequenced by a mechanically rotating commutator. In the Select-a-Rhythm, however, computer logic consisting of a six-bit flip-flop counter and logic gates were used. The tempo of the rhythms could be set to a specific speed, or made to follow the organist by adjusting to the pedal rhythm. In the latter case the voltage produced by the pedal repetition rate was integrated and sent to control a unijunction transistor oscillator [25].

Other organ manufacturers have followed suit and produced rhythm band boxes of various sorts. The Hammond Rhythm can produce cool-jazz backgrounds, slow and hard rock, waltzes, latins, and marches.

Lowry has a special effect which is called Automatic Orchestra Control. This facility sounds notes on the upper manual corresponding to those held down on the lower manual, as the solo manual is played. The notes which are added, sound in the octave range of the notes played on the upper manual. Arpeggios of notes held on the lower manual may be played by either running a finger up the keys of the solo manual, or along a strip of specially prepared switches in front of the lower manual.

Hanert at Hammond Organ developed an accessory which produces a muscial accompaniment to a humming voice. A person hums a melody into a tube and an electronic tone generator of the tempered musical scale is turned on which is nearest to the vocal pitch. The audio signal is then filtered and a vibrato applied [25]. On the Hammond organ, stops are also available to produce the sounds of a honky-tonk piano, harpsichord, big band wa-wa, sitar, and music box.

Computer technology is being used in modern organs to accomplish stop and combination settings. The Kinsman Stop Programmer allowed an organist to insert a punched card which determined the stops which were automatically set by the general action. The Rodgers organ used latching relays to store the state of the combination action. The relays could be simply reprogrammed by depressing the action button and setting the desired stop tablets. Later Rodgers used magnetic core memory to store this state information. Saville has designed a stop and combination action which uses integrated circuit memory to store the stop and coupler state information.

An area of control which could be technically accomplished on electronic organs is touch responsive keying. Baldwin used a type of responsive keying with flexible silver strips which rolled along resistors of logarithmic proportions. As the key was pressed the resistance determined by the point of contact of the silver strip on the resistor decreased [32]. Velocity sensors and pressure sensing devices could be attached to the keys, and the modulation of sounds could be controlled by variable pressure on the keys and a wiggling of the fingers.

A few months ago, in January 1970, Conn installed the largest installation of pipes for an electronic organ in the Cathedral of Tomorrow in Ohio. The loudspeakers from the organ play into the pipes which perform a degree of formant filtering [38].

Other output modifications include the Hammond spring reverberation unit, Leslie speakers with rotating baffles and other rotating speakers such as the Allen Gyrophonic Projection [40]. Reverberation units and filters are being used along with conventional band instruments, as with the Selmer Varitone Trumpet, to produce new sounds and weird effects.

A division of Thomas Organ produces the Vox, which is a combination six stringed guitar and minature organ. When a string touches a fret, the organ sounds with one of its stop settings which can imitate a church organ, rock organ, clarinet, bagpipes, zither, and banjo (with repeat percussion) [145].

Several semiconductor manufacturers have produced special components for the electronic organ industry. Motorola makes a compact twin-T oscillator which they used to build a small organ in a toy piano capable of a big organ sound [96]. General Instruments and others make frequency divider stages (flip-flops) and field-effect transistor switches which will key audio signals [113]. Wurlitzer has used integrated circuits to produce a miniature 24-pound organ which measures only 4x4x36 inches, with a seven-octave keyboard. Phillips Research Labs in the Netherlands announced in May, 1970, a way of obtaining all frequencies of a desired scale from one oscillating flip-flop and a series of frequency dividers. The idea is as follows: suppose we have an oscillator that gives 8 pulses per millisecond and we wish to have 1,2,3,4 and 5 pulses per millisecond. We divide the 8 to give 4; divide the 4 to get 2; and finally make one more division to get 1 pulse per millisecond. To get the 3 and 5 desired pulses we need only to add the 1 and 2 pulse per ms. together and the 2 and 3 pulse per ms. oscillations together respectively. The only problem is that the pulses of 3 and 5 per ms. do not occur at evenly spaced intervals, so the sound is harsh. The general scheme, then,is to add pulses to get the number corresponding to the frequency of the note desired. The idea suggested by Phillips to smooth out the pulses is to begin with a frequency, say 2^8 times higher than before,and then do the divisions and combinations and finally divide out the 2^8 factor. The oscillator used by them to generate an entire gamut of pitches oscillates at around 2 million cycles per second [117].

A final interesting application of computer technology in musical instruments is in the Logical Bassoon. This instrument is rectangular in shape rather than round, and it contains logic circuitry to simplify fingering. There are very difficult-to-finger bassoon passages in well known music, such as Stravinsky's "Rites of Spring," and the micro switch keyboard which drives solonoids through DTL (diode-transistor logic) facilitates easy management of the awkward passages [17].

Electronically Controlled Synthesizers

The fourth and final period of electronic musical instruments overlaps the third period, but includes a different type of instrument. Those in the fourth group are synthesizer-type devices which are controlled by external media such as punched paper tape and analog control voltages.

In 1955 Harry Olson at RCA produced his sound synthesizer MARK I which consisted of several racks of equipment containing oscillators, filters of all types, wave shapers and controllable amplifiers. The idea was to produce an instrument which would be capable of synthesizing any musical sounds possible. Sound parameters were specified on punched 15-inch wide paper that moved at variable rates from 2 to 8 inches per second. The parameters determined frequencies from the 96 pitches possible (F_0# to F_8), the attack and decay envelopes, the filters to be used, and the amplification levels. A disk recorder was synchronized with the paper tape movement, and the resulting synthesized sound could be recorded on one of the six 3-minute concentric tracks on a disk. The mixing of sounds was done by simultaneously playing more than one track at a time and recording the result [116]. Flexible brushes made contact with a drum through the holes in the paper such that continual contact was made for consecutive holes, so that durations of the parameters in time were determined by the number of consecutively punched holes in a column and the speed of the tape. The MARK I had twelve tuning fork tone generators, frequency dividing and multiplication circuits, and glissando-producing circuitry which consisted of a signal integrator and low pass filter [31]. At its debut the synthesizer performed a number of familiar songs including a medly of Stephen Foster tunes, a Bach Fugue and "Nola." The sounds of a piano and harpsichord were demonstrated in the medly, and fugue and "Nola" sounded like a big band arrangement. Dr. Olsen claimed that it was theoretically possible to synthesize Caruso's voice with the background of a symphony orchestra on the synthesizer [60].

The first version of the synthesizer was found to be inadequate and so a further costly venture of around a quarter of a million dollars was undertaken that produced the MARK II version, which now resides at the Columbia-Princeton Electronic Music Center in New York City. Milton Babbitt has done a great deal of composition and synthesis with the MARK II, and he said, "If you know how to specify it, any sound that can be passed by a loudspeaker can be created by this machine" [1]. The RCA synthesizer at Columbia consists of nine racks of equipment, four of which are identical. The MARK II is essentially four synthesizers in one. There is the capability of producing four simultaneous monophonic lines of music by setting the appropriate controls on the identical modules and running the two paper tape drives. The sound parameters are specified either on punched paper or by toggle switches which correspond one-to-one with the punched holes. The punched holes specify 10 columns of binary-encoded information as is shown in Figure 2-10.

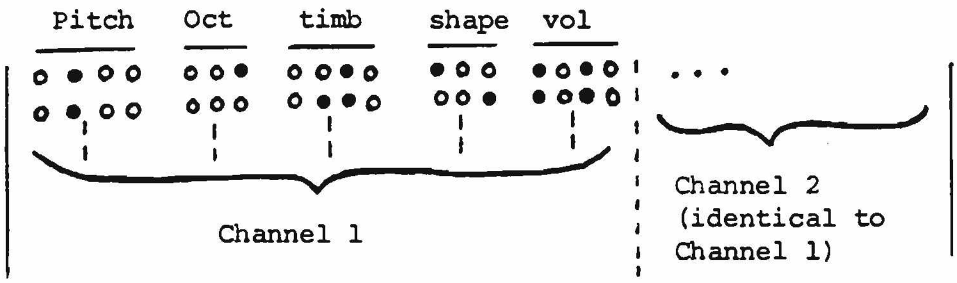


Figure 2-10
RCA Synthesizer Tape

The columns of binary-encoded information which determine relay tree settings are usually organized such that the parameters, pitch (one of the notes, C, C# ... B), octave, timbre, wave shape, and volume are denoted, as shown in Figure 2-10. The three respective single columns of information under shape specify attack, duration, and decay. The synthesizer is very versatile, and the columns can be made (by patch-cord connections) to correspond to other parameters. The paper tape specifies two channels, so two independent monophonic melodies can be described thereon. The punched holes specify relay tree settings, and these settings can be manually achieved by setting corresponding toggle switches. A common procedure for programming is to play around with the toggle switches until a desired sound is heard, then to manually punch corresponding holes by means of keys which directly punch the holes in the paper. There are twenty-four variable frequency oscillators and a noise generator in addition to the twelve electrically driven tuning forks which provide the tone sources. Octavers double the frequency of an input signal and dividers halve the frequencies. Attack and decay times are determined by the envelope generator. The envelope generator is composed of a series of R-C networks with different time constants which range in attack times from 1 millisecond to 2 seconds and decay times, 4 milliseconds to 19 seconds. There are two paper drives, each of which can be synchronized with a multiple channel taperecorder. A signal on the magnetic tape can photoelectrically turn on one or both of the drives. The synthesizer can, of course, act as a signal modifier of any waveform whether produced by the oscillators, fed in live or recorded. The design was so modular that patch cords can be used to route a signal through any of the modules in any order [4].

Both additive and subtractive synthesis are possible on the RCA equipment. At the University of Illinois James Beauchamp developed the Harmonic Tone Generator which allows the additive synthesis of sounds with up to six overtones. With this device one can build up a sound from overtones where each partial can be independently controlled in amplitude and phase. At MIT it was found that individual amplitudes of partials in the attack period of a note are significant in determining the timbre of the tone [74]. The Harmonic Tone Generator was developed to experiment with sounds having dynamically varying partials. The device consists of a series of band pass filters which divides the input signal into distinct frequency ranges. Each of the resulting signals is sent to a voltage-controlled amplifier which determines the amplitude of the partial, and then the signals are recombined. Separate signals can also be fed into the different band pass filters, but one common set up is to feed noise in and vary the volume levels of the separated signals. The instrument has been used for time envelope synthesis and as a voltage-controlled resonator with variable frequency center and bandwidth [67].

In the mid 1960's Robert A. Moog produced his sound-processing devices which allow controlled analog formation of sounds. His equipment consists of voltage-controlled (v.c.) oscillators; v.c. filters; v.c. amplifiers; various fixed low pass, high pass, and band pass filters; white noise generators, envelope generators, envelope followers; keyboard and linear controllers; and multichannel mixers. The voltage-controlled oscillators, filters and amplifiers may be simply and rapidly regulated by analog control signals to generate a certain frequency, pass on a certain frequency response and determine a definite amplitude level, respectively. The v.c. devices borrow their design from analog computer technology [105].

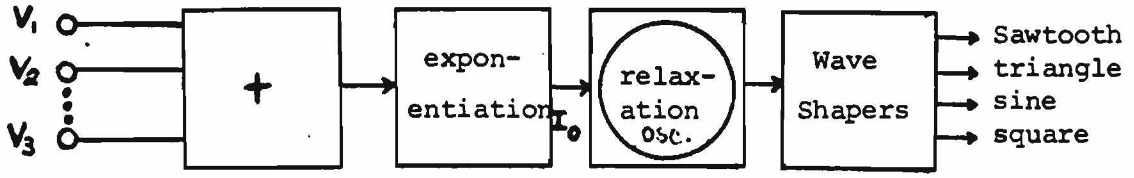


Figure 2-11

Voltage-Controlled Oscillator

In musical pitches, frequency ratios are important; for instance, octave notes are in the ratio 1:2. The v.c. oscillators are built as shown in Figure 2-11 such that an incremental change in control voltage causes a corresponding ratio variation in the frequency output. For example, a one volt rise in voltage causes a doubling of frequency, or a rise of one octave. In Figure 2-11 the current Io controls the frequency of oscillation of the relaxation oscillator. Io results from the exponential sum of the input control voltages: $Io=e^{k\Sigma v_i}$. Io changes in ratio for a given increment in control voltage, and the voltage across a capacitor in the relaxation oscillator is a sawtooth having a frequency proportional to Io. In a like manner the gain of a v.c. amplifier is made to change in ratio with an incremental change in control voltage. One volt corresponds to an intensity change of 12dB and this correspondence holds over an 80dB range [37]. In v.c. filters, characteristics such as cutoff frequency, center frequency response and bandwidth are determinable by voltage control. The following is a list of Moog-type devices showing the inputs and outputs. P is the program signal, or the audio signal being modified and C is a voltage control signal which may range from d.c. to any analog signal.

Device	Inputs	Outputs
oscillator	C	P
amplifier	C, P	P
filter	C, P	P
white noise	none	P
reverberation	P	P
mixer	P	P
keyboard & linear controller	manual	C
sequencer	C, Trigger	C
envelope follower	P	C

Figure 2-12

Moog Synthesizer I/O

Each sound processing module is compatible with all others, that is the input and output impedances match up and the input and output signals have compatible levels. This means that the modules may be interconnected in a variety of ways:

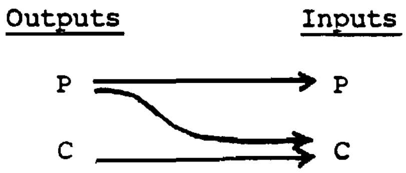


Figure 2-13

Hookup of Compatible Synthesizer Modules

Any signal can be used as a control voltage and any program signal output can be connected to any program signal input.

Another company besides the R.A. Moog Corporation that makes independent voltage-controlled synthesizer modules is Buchla Associates in California. Both systems have patch panels where the various modules are interconnected. Keyboards and linear controllers (a wire stretched over a resistance bar) generate control voltages which may be used to regulate pitch, timbre or amplitude of the sound being produced. A resetting of the patch cords determines what is being controlled. A typical set-up would be to have the keyboard control pitches and the linear controller (with the other hand simultaneously) to control timbre or volume. The flexibility of the synthesizer equipment in imitating sounds is shown on Carlos' record, "Switched On Bach," All the sounds on the record were produced with synthesizer equipment, but not in real time. The keyboard and linear controller are monophonic in that they deliver up a single control voltage at any given time. To obtain a repetitive background voice a sequencer can be programmed to produce a repeated series of control voltages. The sequencer consists of a ring counter which cycles through several stages, each of which outputs a control voltage determined by an individual potentiometer setting. On the Buchla sequencer one can adjust the pulse length and period of the sequence. The length of time spent at each stage before proceeding on to the next is determined by separate knob settings. An interesting combination is to control the center of a bandpass filter with the program signal also being filtered. The result is that the center of the bandpass filter follows the incoming frequency, so each note whether low or high receives the same amount of filtering.

Electronic music studios now, in addition to the classical equipment, are incorporating v.c. synthesizer modules. The normal studio equipment includes audio-signal and noise generators, filters, modulators, amplifiers, reverberation units, mixers, tape recorders, patch boards or signal routing panels, and sound equipment including power amplifiers and speakers [107,132]. An interesting device is a ring modulator which combines two signals to form a third which is a combination of the sum and difference of the two inputs.

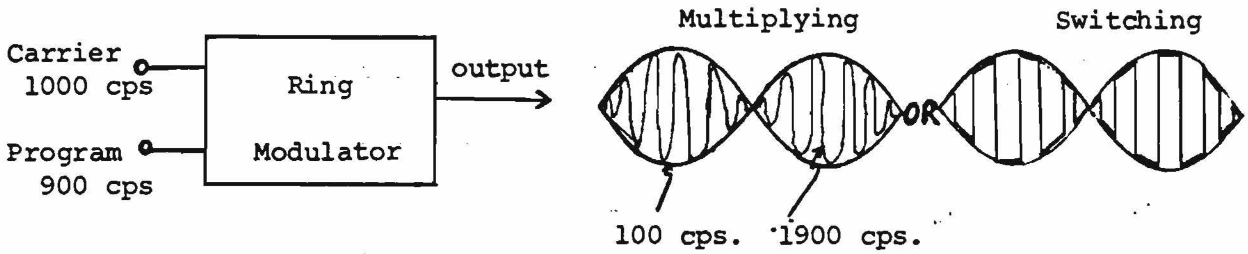


Figure 2-14
Output of a Ring Modulator

There are two types of ring modulators, a switching one and a multiplicative one as shown in Figure 2-14. A ring modulator may be used to apply vibrato to a signal, impart an envelope to a program signal, or generate a spectrum of bandwidth 2n around a specific frequency, m, where n is the bandwidth of the carrier input and m is the frequency of the program signal [13]:

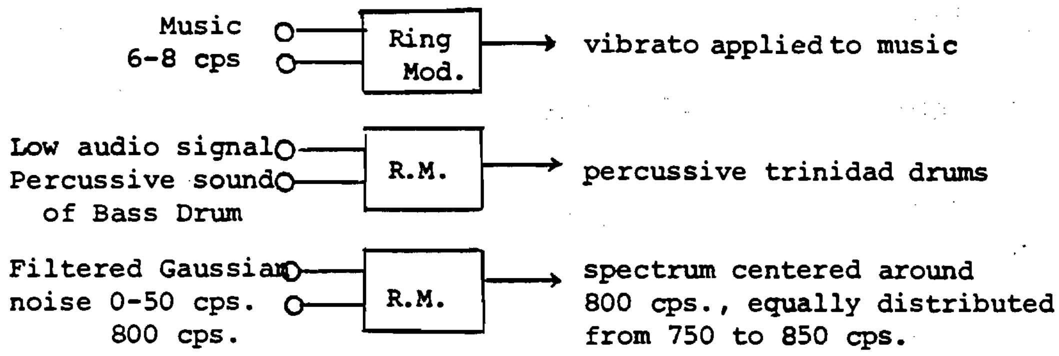


Figure 2-15

Operations with a Ring Modulator

Harold Bode, in a studio at Wurlitzer Organ Corporation, analyzed sounds with electronic equipment consisting of complex tone generators, envelope shapers, filters, ring modulators, percussion gating circuits, dividers, mixers and tape loops. Sounds were synthesized by varying parameters of the equipment and then comparing the results with natural sounds by ear [11].

In East Germany Gerhard Steinke developed the Subharchord which shapes tones with a ring modulator and filters, and then a decay characteristic is rhythmically added to the sound [138]. The three octave keyboard is touch sensitive and the pressure on the key activates the volume. Pressure responsive and velocity sensing keys facilitate the generation of percussive control [110].

In 1965 Paul Ketoff from Rome, Italy presented his Synket which is an electronic system which both generates and controls sound. Like the Moog and Buchla equipment the Synket can be played in real time without prerecording [83]. The instrument consists of a v.c. square wave generator which can be regulated from a keyboard or another source of control voltage. Three frequency divider stages are followed by a v.c. volume modulator which can be randomly varied by noise input. The program signal may then be routed through a variable filter whose bandwidth and center frequency are voltage controllable. A patch board allows the signal paths to be varied at the discretion of the performer. The keys are touch responsive and control both volume and vibrato [37].

Dudley's Vocoder has been used by several people to modulate sound. The incoming sound is broken up into frequency bands by the bandpass filters and the resulting signals may be variously modified by a second sound source such as voice, then re-combined. Alvin Lucier used a Sylvania vocoder by adding and deleting three bands of arbitrarily divided speech spectra to produce his work the "North American Time Capsule 1967." In this instance the human voice furnished both the source and the control signal [89].

New methods of sound production control are being investigated. For example Lucier has tried an electroencephalogram approach at Brandeis University. He attached electrodes to his head and blinked his eyes. The resulting brain waves controlled the sound, which turned out to sound like random noise [131].

Automatic methods of control are beginning to take hold in electronic music production. Emanuel Ghent created his Coordinome, which is a variable-speed punched paper tape reader that is driven by a voltage controlled motor. The control is accomplished through the switching of eight mercury-wetted relays. Twenty-five to two hundred characters a second transfer rate can be accomplished by the variable tape speed of 2.5 to 20 inches per second. The Coordinome was created to provide precise coordination of performers playing electronic music scores, synchronization of live players and recordings, and direction of spacial effects [61].

At the University of Toronto under Hugh Le Caine a major effort has begun to provide programmable control of sound processing equipment. One such device is the Hamograph which modifies sound signals according to a tape loop that has shaped metallic foil strips which are sensed by six channels as shown in Figure 2-16 [129].

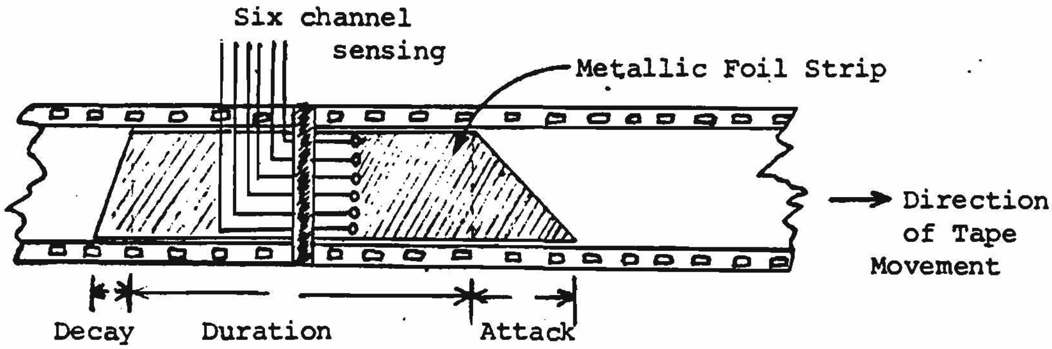


Figure 2-16
The Hamograph

The speed of the tape, which is variable, and the spacing of the foil cutouts determine the rhythmic patterns. The six channels act as voltage sources, and as such can control any v.c. device. Thus they may perform the function of waveshaping (shown in Figure 2-16), synchronization of equipment or coordination of events in time [9,128].

Le Caine has also designed a voltage-controlled tape recorder which has twenty channels, linear controlled level mixers, and ten variable speed tape drives. The linear control mixer consists of a

CHAPTER 3

COMPUTERS AND MUSIC

There are seven areas of involvement of computers in music which are : 1) generation of sound waveform samples, 2) music composition, 3) music printing, 4) music analysis, 5) musicology, 6) education, and 7) control of music-generating media.

Generation

Max Mathews, in his book The Technology of Computer Music [69] describes his method of computer generation of music. Essentially his program, MUSIC IV, builds up sound waveform samples from waveform parameters defined either by the user or by a special routine. At the output the waveform samples are converted to an analog signal by passing them through a digital-to-analog converter and low pass filter. To accomplish a frequency response of N in the analog signal, 2N samples per second must be generated, so on the order of 20,000 to 30,000 samples per second are usually calculated. The specification of complex waveforms may cause as much as a factor of 30 to 50 times real time to generate the desired samples [118]. Complete generality is obtained with this approach, but it is very costly in terms of computer time and money. Another disadvantage is that the user must wait until his analog tape is produced before he can hear the sound. The Music IV program permits the definition of several "instruments" which consist of variously connected waveform generation and modification routines. Once a set of instruments has been defined, they can be addressed by note-card parameters which determine the duration, intensity, frequency, and time a certain instrument is to play. A symbolic representation of the function generators allows the connection of generators and function modifiers in a flowchart form which is simple to follow and conceptualize [94]. All conceivable combinations of functions can be specified to effect periodic and non-periodic waveform generation; addition and multiplication of waveforms; formant, low pass, high pass and band pass filtering of waveforms; controlled and random modulation of frequency, pitch, filter bandwidth, and filter center frequency; and precise generation of attack characteristics, duration fluctuations and decay parameters [142].

The evolution of Mathew's music programs is very interesting. MUSIC I started out with a single voice and one fixed waveform. MUSIC II had four voices and a choice from sixteen waveforms but no attack or decay. MUSIC III had the unit generators with the idea of one oscillator controlling another oscillator, etc. and the definition of instruments. MUSIC IV was essentially the same as MUSIC III with better computational facilities which included a Macro Assembler especially developed for the music project at Bell Labs. MUSIC V is generally the same as MUSIC IV (except for auxilliary routines added by other users of the program). MUSIC V is coded mostly in FORTRAN except for some tight machine coding of the unit generators, thus it is more easily transferable from one machine to another. An informal MUSIC V user society has been formed among many users in various universities who are working with variations of this program. Tuck Howe and Barry Vercoe at Princeton under the direction of Professor Winham have been actively pursuing futher developments to MUSIC V.

In 1965 Arthur Roberts developed his MUSIC 4F music-generating program which was an adaptation of MUSIC IV on the Argonne National Labs CDC 3600 computer. He accumulated a library of "instruments" and had provisions for using any thirty of these at a time [124]. The definition of his instrument #1 was so versatile that it approached the generality of the entire orchestra compiler of MUSIC IV. For example, player #1 had the choice of several waveforms including sawtooth, sine, square, and variable bandwidth and formant-modulated sawtooth, and wave shaping capabilities including envelope attack and decay curve definitions, vibrato and tremulant settings and the selection of filtering by any of thirty arbitrary filters. A smaller version of this program was put on a small computer at Argonne by Robert Clark. He called his program MAESTRO. With this program the user defined up to nine different musical instruments and supplied the inputs of frequency and amplitude to control each one. A query program assisted the user in defining an instrument by asking for parameter settings for each instrument in an interactive way. The user could also define a scale by specifying the number of intervals per octave, and he could also indicate the desired tempo in beats per minute. The "instruments" could be played in real time since the sample rate was reduced to 6000 samples per second which gave a frequency response of 3000 cycles per second [20].

Composition

Computers can be used in essentially two ways in music composition. The first is the direct composition, organization and structuring of musical works by the computer according to a set of specified rules and conditions. The second is an interactive role in which the computer serves to store and retrieve musical passages, suggest possible musical combinations to the composer, and perform harmonizations and structuring of passages under user direction. There are degrees of direct computer composition ranging from a highly random selection of pitches, rhythm and orchestration to the highly specified determination of parameters. Mozart wrote of a dice game which he used to compose his Musiklisches Wurfelspiel ("A Musical Dice Game"). In 1955 this chance compositional procedure which ordered assorted passages of music according to thrown dice, was programmed on a computer [72].

The idea of having a computer compose in the tradition of classicism, Romanticism, or a specific composer has intrigued many computer composers, and an information theoretic approach has been attempted by many people to realize this result. The idea is that a style of music contains a certain amount of information entropy and that if the information content and degree of entropy can be discovered and modeled, then production of Brand X music can be realized. Entropy is the degree of randomness or disorder in a physical system, and information entropy is the measure of content. Information content is not synonymous with meaning, but rather with the amount of information that could be contained. In other words, entropy of information relates to what you could say, not what you do say.

In 1961, Harry Olson and Herbert Belar reported use of transitional probabilities in their composing machine. They organized a table containing the frequency counts of pitches in eleven Stephen Foster songs. Then they built first and second order transitional probability tables and noticed that there was redundancy in the tables, and thus a definite pattern to Foster's music. They wired a probability matrix based on the tables and randomly selected pitches according to these weighted probabilities. Each note was determined by random number-generating oscillators with a probability depending upon the preceding notes. Weighted choices were also made of the rhythm structures of each measure. The melodies produced by their device which had the capability of only twelve output notes, sounded very much like Foster tunes [115].

Another similar approach was taken by Brooks and his associates to analyze and synthesize tables. Tables up to 8th order were built up from the statistical analysis of 37 hymns which were all normalized to a common base, the key of C. The computer program generated random numbers which were accepted or rejected in accordance to weighted probabilities. Their results showed that low order of synthesis produced a non-typical example, a middle order of synthesis produced an example which belonged to the class, and high order generated much duplication, in whole or part of the original sample [73].

At the University of Illinois while Lejaren A. Hiller was at the helm, there was a great interest and work done with Stochastic models, information theory and statistical analysis in the generation of computer music. The Illiac Suit, done on the Illiac II computer, was one of the several compositions that resulted from this work. The different movements were composed by techniques based upon varied models. The first movement was based on the rules of classical counterpoint such as stepwise motion following a skip, no parallel unisons, octaves, fourths and fifths, no dissonant intervals, no skips over a certain range, etc. A cantus firmus (reference melody) was generated at random with rejection of those notes that violated any rule and then the counter melodies were developed. A number was generated at random, and if it represented a legal note, then it was added to the development. The second movement demonstrated random generation of chromatic music. Here many of the rules such as no dissonant chords, and no parallel fourths and fifths were relaxed and a rule of restricted tone rows was added. The music produced in this section had higher information content (was more random) than the first movement. The third movement consisted of musical textures based upon stochastic models and Markoff chains. Here the entropy was very high [70]. At one intended performance of the Illiac Suit the engaged string quartet fled from the stage, declaring that the score was unplayable.

Other stochastic compositional experiments were carried out with the aid of MUSICOMP, a macro oriented language which allowed the use of several statistical analysis routines, weighted probability random number generators and twelve tone row permutations. Arnold Schonberg's 12-tone techniques were especially suited for computer programming and synthesis. In one experiment a set of strict rules was set up which represented the 12-tone techniques; then these were fed one by one to the sclection routine which determined the note sequence. Random numbers were generated and screened by the rules. The music began as white noise, and as more and more rules were put in force, the music settled down to a degree of predictability [72]. Divilbiss proposed the coupling of two computers, one composing the music and the other one synthesizing it. The question, "Does what I've composed exceed the capabilities of the performers?", has been replaced by the question, "Does what I've composed exceed the capacities of the listeners?" [24].

Printing

The printing or writing of scores has been done in several ways with computer assistance. At the University of Illinois musical scores were plotted on a Calcomp plotter. In one version the printed output represented a plot of dynamic level versus time. It was claimed that a performer could play directly from the score after a short introduction [73]. J. Divilbiss wrote a routine for the Calcomp plotter to produce a player piano roll. The plotter marked places on the roll where the holes should be and then the holes were punched by hand.

At the University of Colorado, Cecil Effinger modified an ordinary typewriter with a music notation type font and added the capability of overstrikes for characters that occur at the same place such as note head, stem, and beginning of a tie. In cooperation with Mr. Effinger, L. A. Hiller and R. Baker adapted an automatic control to the mechanical music typewriter and controlled it with a 5-hole paper tape. Musical scores were produced and the following conclusions were reached: the preparation of scores was faster than by hand; there were not enough characters available even with the case shift; various sized and oriented symbols were needed but could not be produced, and the spacing tolerances were difficult to maintain.

Max Mathews has produced printed scores in the form of graphs. He and Miller produced a collection of four graphs versus time depicting log frequency (pitch), loudness, duration, and duty factor (staccato or slur). Mathews and Chang have also worked on other graph-type score drawing programs which performed automatic scaling of amplitude depending on the maximum pitch range of the voice being plotted [21].

Analysis

There are two main aspects of computer analysis of music. They are the analysis of sounds and the analysis of music compositions. Analysis of natural sounds is interesting from the physical and physiological points of view. That is, from the aspect of understanding more completely the physics involved in the production of a sound and from the view of learning how a person perceives and processes sound. It turns out that the mere shape of a waveform on an oscilloscope is not sufficient to determine a sound fully. For instance two violins playing the same tone may give identical waveforms, but the sound of one could be that of a Stradivarius violin which sounds much different from that of the other which could be a student instrument.

Fourier analysis of sound waves has been the general technique used by most people. Glenn Culler found that a non-Fourier analysis of speech waveforms gave better representation of speech synthesis parameters. At the Universtiy of Illinois David Freedman developed a non-Fourier analysis technique which began with the representation of waveforms similar to that of Culler, namely $f(t) = \sum_{k=1}^{m} A_k(f)\mathrm{Sin}[W_k(f)]$, in which the amplitudes and frequencies are time-dependent and the frequency is not necessarily harmonically related. Very life-like violin and wind tones were synthesized in which the violin tone sounded bowed and the wind tones sounded breathy [55].

It has been discovered that we do not understand the sound production or ordinary musical instruments so well as commonly believed [67,66]. Current work is being done at Illinois by J. Beauchamp in the area of forming new mathematical models of physical systems. Being a trombone player, he is working on a new model describing the production of sound on brass instruments using a trombone mouthpiece with various pressure wave sensors attached to it. The resulting waves are compared graphically with those generated on the computer by his proposed mathematical model, and the results are encouraging.

The aspect of computer analysis of music is closely allied with music composition because much analysis has been carried out to determine rules and trends to be used in music composition. Several mathematical models and algorithms have been programmed to facilitate analysis of harmony. A set theoretic approach has been used by Robert Mason to determine harmonic and transpositional equivalence of separate note groups [93]. A linguistic approach has been proposed and outlined by Terry Winograd who used the programming language LISP to construct a primitive grammar and perform harmonic analysis. The idea is to look at different parsings of the melodic lines rather than at simple static chord structures. The context dependence of notes in chords must be discovered in relation to the movement of melodic lines to achieve a higher performance of harmonic analysis [152].

One of the big problems in music analysis is the transcription of the music notation into linear form for the computer. Several proposals have been offered by a number of people. Different degrees of representation have been used; for example, some approaches have consisted of coding only pitch information, others of coding pitch and duration, and others of representing all information available from a written score. The argument in favor of the latter is that even though simple analysis of pitch structure is to be carried out, maybe the first analysis will suggest other interesting relations with other parameters. If the information representing the other parameters is not available in coded form, then recoding must be undertaken.

Allen Forte has proposed a syntactic parsing procedure for the analytic reading of musical scores [152]. The scores are first linearized on punched cards with Stephen Bauer-Mengelberg's notation, then the information is parsed by pattern recognition programs written in SNOBOL3. The musical input is segmented according to instruments and the time they play. Programs have also been written which analyze the parse to reveal similarities and differences of segments, distributional characteristics, transpositional, equivalence and inversion relations and set membership statistics [52]. Bauer-Mengelberg's notation is a highly mnemonic one-to-one correspondence of written music notation into a string of symbols representing note placements in relation to staff lines. The representation is neutral with respect to the pitch assignment of the staff lines [135]. Robison's Intermediate Music Language [47] and Brook's and Lefkoff's "Plaine and Easie Code System for Musicke" [135,18] make pitch assignments directly as input note parameters. The future automatic scanning and reading of printed scores by computers will ease the problem of input coding but will necessitate further work in graphical parsing techniques and pattern matching.

Once the music transcription has been accomplished there are two general approaches to music analysis. The first is the statistical approach in which the composition is considered as an objective specimen in a set. This is the most common method and various statistical parameters have been calculated and compared by Erickson [47], Brender [16], Forte [52], Selleck and Bakeman [134], Fuchs [68], and others. The second approach is the information theory approach which was suggested by Hiller at the University of Illinois. The goal of this method is to find a relation of information content and music and psychological response. Information measures such as bits per symbol and bits per second are studied and compared [68].

Attempts have been made to compare statistical and information theory analyses [69]. Objections have been raised to the information theoretic approach on the grounds that information theory is only applicable to Markovian processes and that musical melodic lines cannot be modeled even by higher order Markov chains because melodies vary freely in length. A new linguistic analysis approach has been suggested which looks at generative techniques with grammars that are self-embedding and conform to Chomsky-type transformation rules [135]. Essentially the problem of analysis boils down to how contextual interrelations of the various music parameters can be expressed and determined. Computer-science and linguistic research of transformational grammars is beginning to be directed toward the problem of music analysis and some interesting results may be forthcoming with this approach.

Musicology

Computer-aided musicology is concerned with the information storage and retrieval of music bibliographies, music history and research, and thematic indexes. The largest project of this kind is the Repertoire International de la Litterature Musicale at Queens College under the direction of Barry Brook and Gary Berlind [122]. The goal of this project is to categorize then classify all scholarly articles, books and papers in the area of music, to provide a comprehensive thematic index of serious music and to organize compositions in relation to analysis of structure, style, harmonic content and melodic movement [18]. There is a recent trend of musicians using the computer to aid stylistic studies and categorization of various forms of music ranging from modern atonal music to folk songs, and back through the ages to Gregorian chants and 12th century music [15, 48].

Education

Electronic music has made its way into several elementary, junior high, and high schools, and the near future will see the computer being used to help generate the new musical sounds [80]. Computer-aided Instruction (CAI) in music has begun [122], but this area will see much more emphasis in the near future as computers are interfaced with music keyboards and display terminals. Two successful programs in music education deal with fundamentals of music and aural perception. The first of these consists of an interactive part-writing program which instructs the students in the basics of music theory, chord choice and movement, and melodic line harmonization. In a control study of part-writing, students who used the computer approach spent a little more time working than the others in the class, but they produced much better musical scores, as judged by the professor of the course [49].

The aural perception experiment was done by Kuhn and Allvin at Stanford University to teach sight singing. An analog sampling device measured the fundamental period of the sound being sung, and this information was used by a computer to extract pitch. The computer then indicated to the singer how close to a designated pitch he was singing. Typically, the computer presented a series of notes to be sung, and the person would sing them into a microphone. If the note sung was not within a pre-specified tolerance, then the computer indicated how much below or above the correct pitch was being sung. The student could choose the tolerance and whether or not he wanted to have the pitch sounded.

Programmed music teaching is primarily being done with films, pictures, tapes and accompanying text books. The student uses buttons to control the movement and takes dictation from the tapes. In a computer-controlled environment the student has immediate access possibility to a large bank of stored data and stored procedures. The future of computer-aided learning of music has just begun and should be exciting.

Control

Computer control of music processing media can be divided up into the following groups: control of analog equipment, control of digital waveform generation, and control of a digital switching matrix.

An area receiving a lot of current attention is the computer control of the voltage controlled synthesizer modules. The idea here is to let a small computer run the synthesizer by interpreting a simple composer notation [80]. All types of voltage-controlled equipment can be controlled by a computer with a D/A converter. A suggested set up at the University of Illinois is to have several control channels that are duplexed from a few D/A converters. In this manner the fast digital computer can service several control devices in real time [54]. At the University of Toronto nearly all of their bizarre equipment could be interfaced by a central computer that would control device interconnections as well as individual control inputs. The individual control inputs determine rhythm, mixing, pitch, attack, and decay characteristics, dynamic levels, timbre, and special effects [58,129]. The central problem in real-time electronic music generation is that of interactive control of the various music and sound parameters.

Computers will help in the future by assuming the role of specifying automatically those control parameters which are fixed or preprogrammed, and providing the user with interactive response to those sound parameters which the user chooses to control. At the Bell Laboratories Max Mathews is using digitally converted analog signals to control electronic circuits which simulate stringed instruments (violin, cello, and bass violin) very well.

The computer control of digital waveform production is done by Max Mathews in his latest work with the programming system GROOVE. The idea here is to provide capabilities for the editing of time functions. The system is operative on a Honeywell DDP 224, 32-bit word computer with a large 8 million word secondary disk storage. Time functions are described by a sequence of numbers representing a period. As many as fourteen functions may be produced for several hours with up to 200 samples per second. When a sampling time interrupt comes in, the functions for the current time are looked up or computed by linear interpolation and sent to the output. The resulting functions can be used as control functions to regulate any v.c. equipment. Manual control devices which input information into the computer consist of a small organ keyboard, a set of potentiometers, and a three-dimensional wand. These inputs are used to digitally control the production of the time functions which in turn may control analog equipment. A neighboring display scope has been programmed to display any of the defined time functions in real time, or in slow motion, with time under direct observation and supervision of the user. By using this system the programmer can act as conductor directing control while leaving the individual tone production up to the various components [95].

The final aspect of computer control is the direct digital control of a matrix of switches which perform the digital function of enabling or disabling a signal path. This method of control is used in our project to control signal paths of the organ tones and filters. In our case, we can achieve programmed direction of several polyphonic melodies each independently controllable in voicing and amplification. At Toronto a series of 20 latching contacts are operated under computer control. These 20 switches correspond to the 20 channels on their special stereo taperecorder [57]. A fully computer-controlled control panel is planned and being designed by J. Beauchamp at the University of Illinois. Here the idea is again to switch audio signals under computer control. This means that the computer is assigned the task of establishing interconnections between equipment. This is very interesting as attested by Max Mathews who has some control of device hook-up at the computer console. When this is done it is a trivial matter to write software which conveniently facilitates the job usually done at a patch board or mixing panel, and which quickly and clearly displays the state of the interconnections.

CHAPTER 4

THE COMPUTER-AIDED MUSIC TOOL

General Description

The Musicational Tool derives its name from three areas: music, communication, and education. The Tool consists of a computer-driven electronic organ. The computer is a communication and control center which directs peripheral equipment including an electronic organ, a CRT display scope and a teletype. The basic idea of the computer-aided music project is to provide computer storage, speed, data manipulation, and program control to help a user manipulate music information. The organ is "full-duplexed" through the computer. The term "full-duplex" is usually applied to a teletype-computer system where the depression of a key sends a specific character code to the computer without mechanically typing the character on paper. The computer receives the code, processes it according to some stored procedure, and sends back information to the teletype as to what should actually be printed. A half-duplex teletype system, on the other hand, is one in which the key depression causes a character to be sent to the computer for processing. In relation to the organ, full-duplex indicates that a depression of the manual keys, pedals or stop tablets sends the positional information to the computer, but it does not cause any tones to sound. The keying busses are disconnected from tone generators so that the keys cannot directly key the notes. This separation is accomplished at only one point by a switch, thus enabling the organ to be used in the normal fashion by the flick of a switch. The computer, upon receiving the information as to which keys have been depressed and which stop settings have been set, can cause several actions to take place depending upon the stored programs. The computer may be programmed to "echo" the input information back to the tone generators and filters, thus making it sound as though the organ tones are being played directly.

We built the electronic organ from kit form in order to become acquainted with the electronics and so that we could modify the organ to be interfaced to the computer. The organ contains two full 61-note manuals, a 32-note pedal clavier and more than 20 stop settings. The organ generators, dividers and filters consist of discrete electronic components. The twelve transistor oscillators have inductance coils which maintain stability and facilitate tuning over an approximate range of two notes. The oscillators are tuned to the equitempered scale from $C\#_7$ (2217.5 cps.) to C_8 (4186.0 cps.). The lower octaves of each note are obtained by the six frequency divider stages per oscillator (seven, in the case of C). Thus the entire pitch range of the organ is 85 tones, C_1 (31.7 cps.) to C_8 (4186.0 cps.). These tones are generally directly keyed onto a series of busses representing 16-foot, 8-foot, and 4-foot, etc., pitches and then fed into the bank of filters. The electronic filters are either grounded out, or switched in the circuit to color the sound. Each filter circuit can be plugged into place so that the replacement or changing of filters is simple. Upon leaving the filter circuits, the signals pass through the preamplifier-vibrato tion on playback. The recorder-modem interface data rate is ten times faster than the teletype speed.

The Organ Interface

The organ-PDP-8 interface is capable of handling around six hundred inputs and outputs. There are 204 inputs to the interface from the manual keys, pedals, and stop tablets, and 396 outputs to the tone generators, lights on the keyboards, and filter switches.

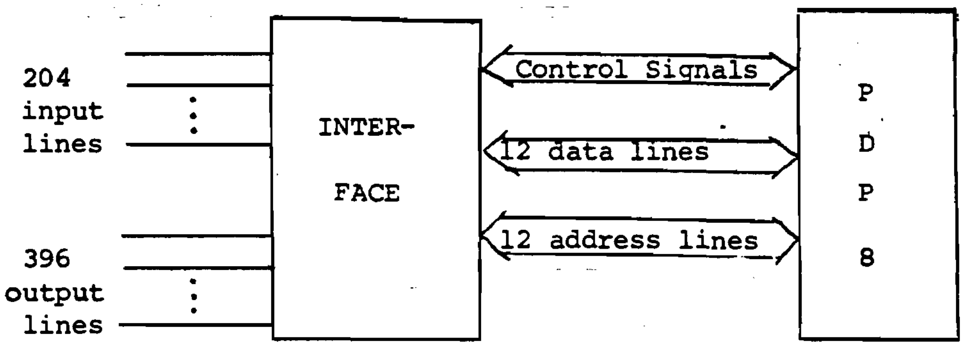


Figure 4-1
Organ-Computer Interface

The 600 I/O lines are divided up into 50 groups, each consisting of 12 lines. These groups correspond to 50 of the PDP-8's possible 64 I/O channels. The computer can address the 50 lines directly and independently by I/O transfer (IOT) commands. On input, 17 of these groups are polled by separate IOT instructions to get the state of the organ into the computer. This data transfer is thus set up so that the data goes through the computer. We originally planned to have the state information go directly into memory with the databreak feature, but we would need a multiplexer to allow us to use more than one device on the databreak channel, and unfortunately we have no multiplexer. The input state information is stored in the computer as a 17 x 12 bit matrix where the 1's indicate a key depression and/or tablet setting "on", etc. The input transfer rate through the accumulator is about ten times slower than that of the databreak rate, so the maximum input rate is around 66,000 words per second.

154 keys (both manuals and pedals
40 stop and coupler tabs
10 vibrato-volume settings
204

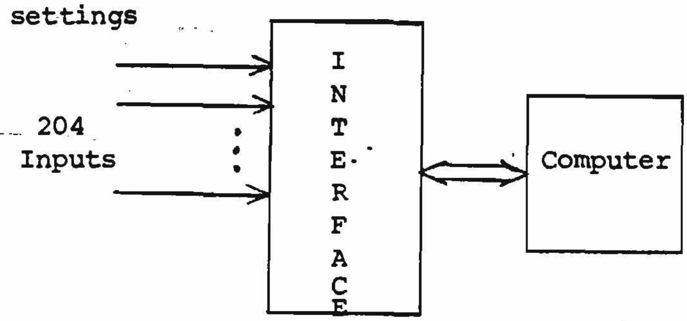


Figure 4-2

Input Information

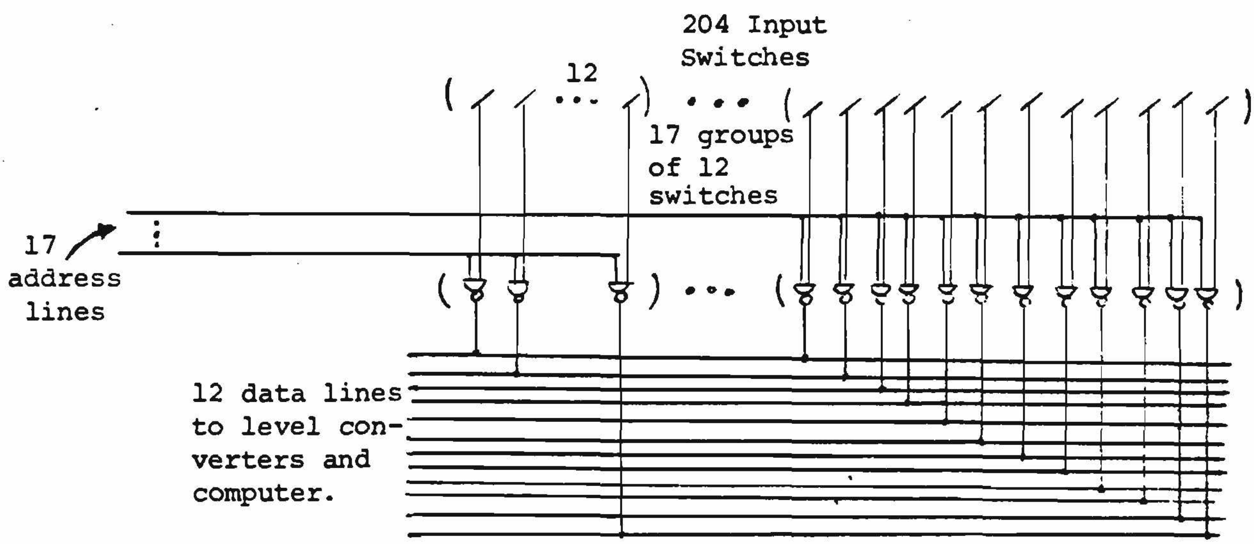


Figure 4-3

Data Input Interface

On input the computer will designate one of the 17 input groups and the appropriate address line is enabled, gating 12 switch states onto the 12 data lines which enter the computer via the level converters. These are necessary to convert the TTL (transistor-transistor-logic) levels of +5 and 0 volts to DEC logic of 0 and -3 volts. The level converters are manufactured at Montana State University under the direction of Dr. Louis Schmittroth.

After the processing of the input information by user programs, the output state matrix is built. This matrix is 33 x 12 bits and contains 1's and 0's corresponding to how the output data switches are to be set. The switches control the tone generators amplifiers, lights and filters.

255 tone switches
122 lights switches
19 filter switches
396

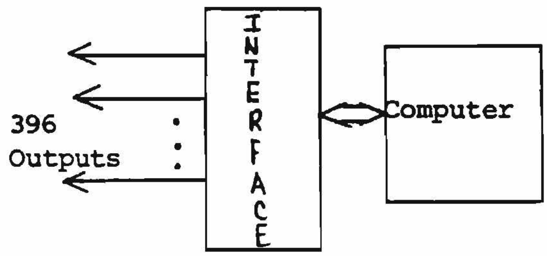


Figure 4-4
Output Information

Associated with each output switch is a latch which stores the state. When a signal is directed to the latching flip flop, it is set to a corresponding state and opens or closes its associated switch.

The switches we chose to switch the audio signals are field-effect transistors (FET's) because they are capable of switching analog signals with little distortion. Essentially an FET acts like a large resistance of around 10^{10} ohms until its gate is biased negatively, then the resistance drops to a few hundred ohms, allowing the audio

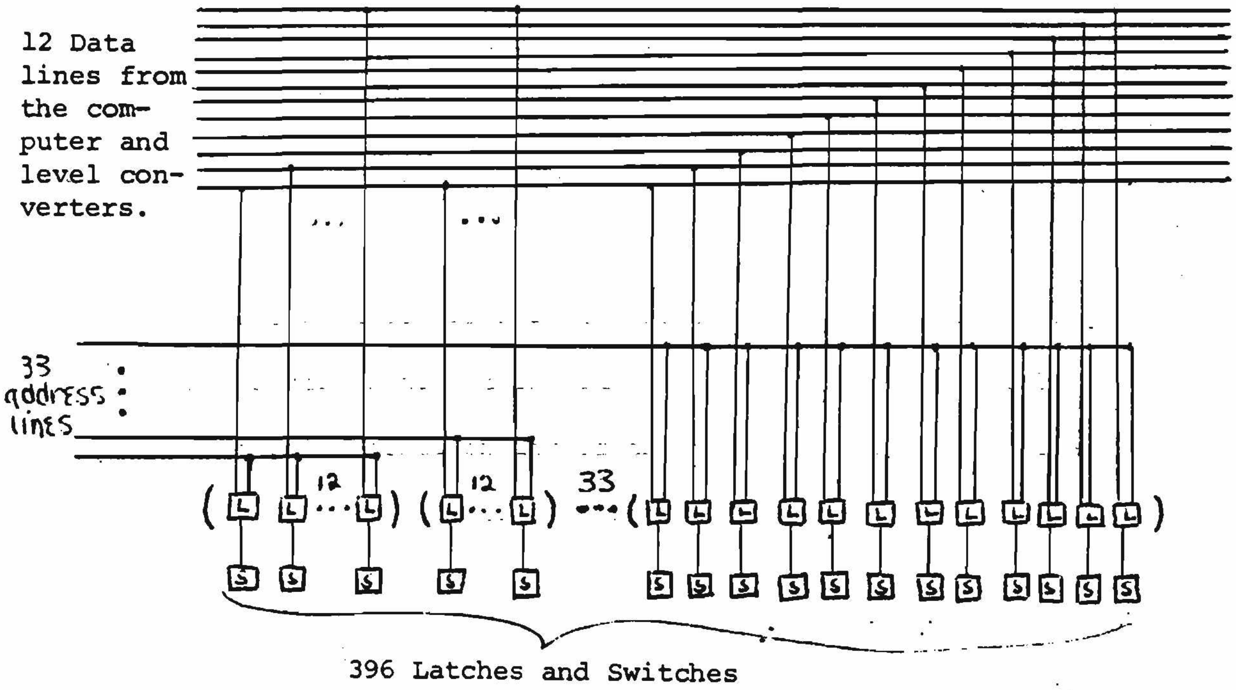


Figure 4-5
Data Output Interface

signal to pass through. The FET's are especially nice since they can be driven with TTL logic levels and are packaged compactly. The integrated circuits we are using contain 10 FET switches per 24 pin package. They are very expensive at this time, but I expect to see integrated FET's become much less expensive in the future due to ever increasing demand because of their versatility in handling analog signals. In the past year the price of these switches dropped about 50 percent. Another approach we could have taken is that of using reed relays for the switches. We experimented briefly with relays, but decided to use the FET's for experience and long-life reliability. We also were not sure if the faint clicking of the many switching relays would create an audible annoyance.

The tone and filter switches are organized as follows:

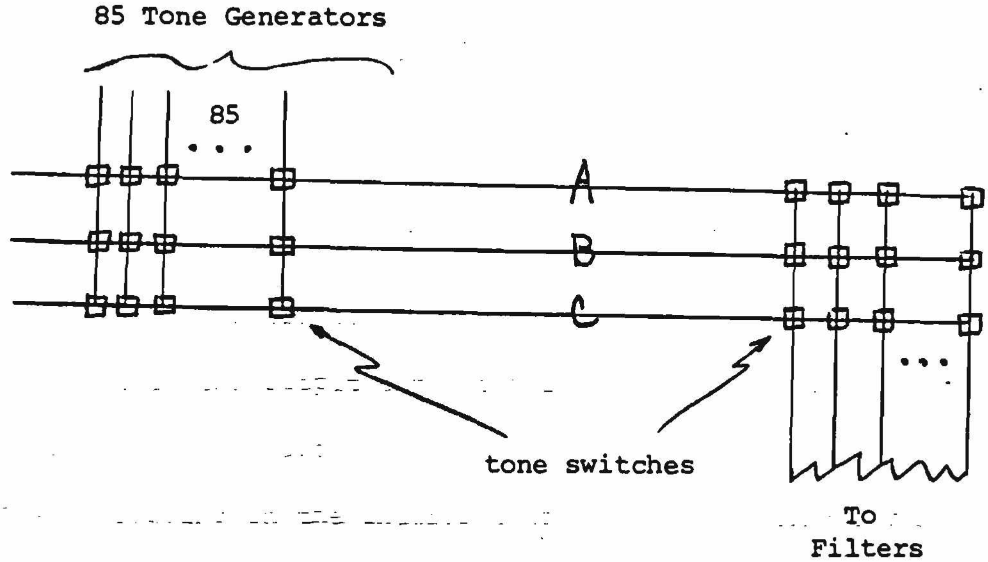


Figure 4-6

Output Tone Switching

The output tone switching is organized into three registration busses--A, B, and C--each of which is able to carry any combinations of the 85 tones. The tones on a particular bus are voiced by the filters which are switched on. Thus the computer is capable of playing three independently orchestrated polyphonic voices. The voices may, for instance, be made to correspond to three manuals with different stop settings and having the entire pitch range of the organ apiece. This is one of the first experiments to deal with interactive computer control of many polyphonic voices simultaneously in real time. The rapid computer control of the filter switches as well as the tone switches facilitates the control of dynamic tonal colorings as various lines progress in time. It will be very interesting to experiment with voicing flow as well as melodic flow in new musical compositions.

One drawback to the design as shown in Figure 4-6 is that the entire range of the pitches are directed to the same filters. Since filters are designed to operate in a specific, limited pitch range, the filtering would not be uniform. The diagram is a simplified logical picture of what actually happens. Each registration bus is broken down into three sections, low, middle, and high. The low sections go to one bank of filters, the middle sections to another bank, and the high sections to a third bank. Any combination of filters in each bank can be switched on any of the three lines of the corresponding section. Voicing filters come in threes, one for each section. Each group of three filters is designed to give a fairly uniform sounding tone over the entire 85 pitches. The usual procedure is to have the correspondingly equivalent filters switched on for all three sections, but it is possible to choose entirely different tone colors for the sections.

An example of how the audio switches are organized is shown in Figure 4-7.

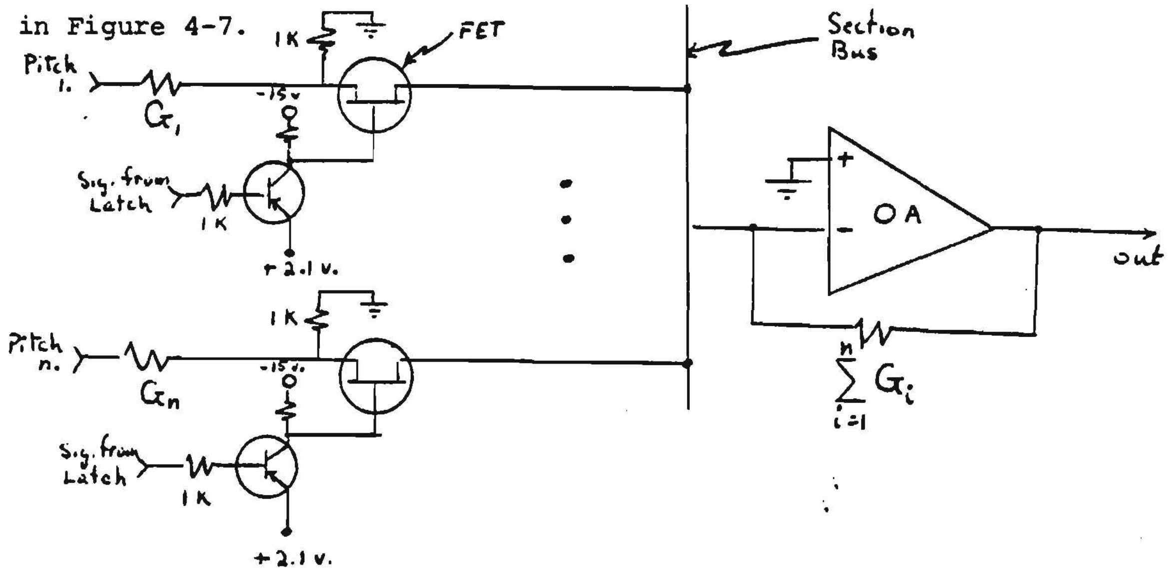


Figure 4-7

Switch Circuitry

As Figure 4-7 depicts, the tones are gated onto the section bus and added by the operational amplifier (OA) circuitry. The signals from the latch flip-flops bias the PNP transistors which drive the FET gates. The purpose of the operational amplifier is to keep the section bus at virtual ground, thus keeping the tone inputs isolated. As the operational amplifier adjusts its feedback to keep the bus at ground, the output circuit varies inversely as the sum of the inputs. Thus the audio signals are added and inverted.

The addressing in the interface is done by decoding a six bit address into one of 64 lines. Actually only 50 lines are used, but it is nice to deal with powers of 2. Seventeen of the address lines are used for input from the organ and 33 are used for output from the computer. A polling program makes IOT requests to the I/O lines and sends a 6-bit code which designates one of the lines. The decoder takes the 6-bit address and puts a pulse on the appropriate address line.

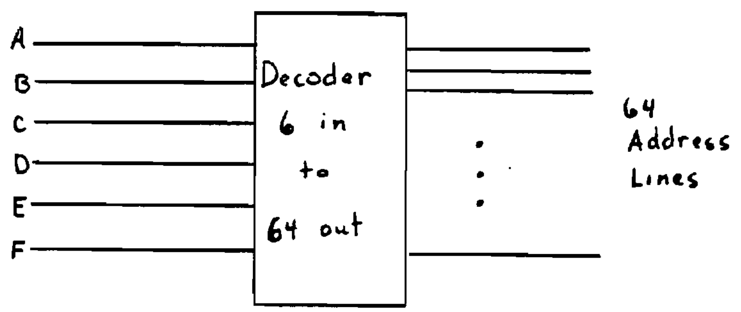


Figure 4-8
Interface Addressing

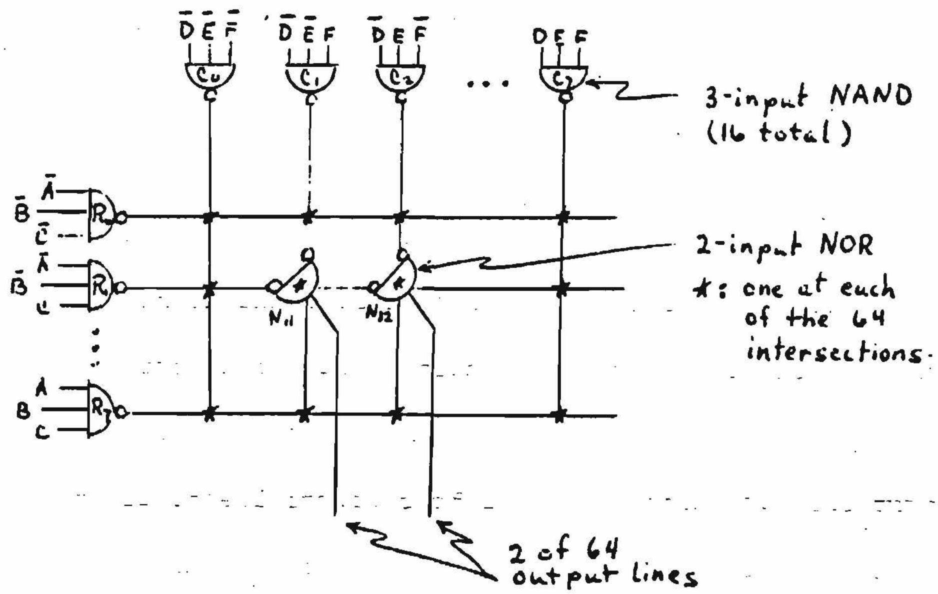


Figure 4-9

Decoder

As shown in Figure 4-9 the decoder is composed of 16 triple input NAND gates and 64 dual input NOR gates. Six inverters are also needed to supply the NOT's of the 6 input signals, A, B, C, D, E, and F. At each of the 64 intersections of the eight-by-eight matrix a dual input NOR combines the appropriate signals to produce the address line. An example of how the decoder works is now given. Suppose the 6-bit address A, B, C, D, E, F were 001010. Referring to Figure 4-9 only the NAND gates R1 (row 1) and C2 (column 2) produce low outputs since only $\overline{A}$, $\overline{B}$, C, $\overline{D}$, E, $\overline{F}$ are high. The two low signals cause the NOR gate N12 to yield a high signal on the corresponding address line. None of the other NOR gates produce a high signal because their inputs all have at least one high signal.

In summary, the I/O of the PDP-8-organ interface works as follows: the computer issues an IOT request to one of 50 lines. The first 17 of these are input lines, and when addressed each one gates 12 bits of information into the computer. Each bit of information represents the state of a manual key, pedal, or stop tablet position. The final 33 address lines are output lines, each one of which gates 12 bits of information from the computer into 12 specific latches which determine the state of 12 corresponding switches. Each of the 396 (12 x 33) switches controls a tone amplifier, light or filter.

Internal Representation

In the representation of information in a computer there is always a trade-off between time and space. For instance, a compact method of coding such as Huffman coding takes up relatively little space, but usually requires extra decoding time. The time-consuming functions, such as decoding and information management can and should be done with special hardware, and then one gets both efficient storage and processing of information. It is with these goals in mind that we have designed the internal representation of our music. As time goes on we of course plan to develop a high-level language which uses our representation scheme as a kernel. The interest of a high-level language in representing information is that fewer bits of information are needed to express the information. Another important idea we are trying to keep in mind in designing our representation language kernel is that routines and packages of information should always be well nested. Control should be restricted so that spurious jumps cannot take place that would cause the execution of some data that was never intended to be programmed. The idea of building systems up by subroutines is also important. Small routines are created and checked out so that they run the right way and larger routines are built up out of the "known" (hopefully) correct small routines. I will come back to these ideas when I talk about the kernel language.

The input information of the state of the organ is stored in the computer, processed by some user-defined programs and finally results in some other information being sent to the output. Mathematically this process can be described in functional notation as $y = f(x)$ where x is the input, y the output, and f is a complicated function which is itself composed of several functions. The functions are procedures which accept and process the input data. The computer routines take the input data and perform various processes and algorithms based on the input information. The output y, though dependent on the input is usually entirely different. For example, on a particular key input, the computer output may turn on several tones, or lights, etc.

The following describes the evolution of our present internal data representation. The first idea was to represent the <u>changes</u> from one time to the next. The development of this idea proceeded as discussed below.

The input to the computer consists of the 17 x 12 bits information describing the state of the manual keys, pedals, and stop settings. This information can be stored in a 17-word array in the computer. This information will be referred to as an input 17-by-12-bit matrix. There were two such state matrices that handled the input. When the input data came in, it was stored in a matrix called the new state matrix. A bit-by-bit exclusive-or was then done between corresponding bits of the new state and old state matrices. Then the change information was stored in a list. The state information of the new state matrix was transferred to the old state matrix, and the process was ready to start once again. Actually, the bits of one matrix were not to be transferred en masse to the other, but instead the roles of the two matrices were interchanged. The exclusive-or function resulted in information describing the changes of one time period to another. The result was a significant reduction of data. For example, it is much more economical to represent the changes of state from one time to another which can usually be done with fewer than five or six

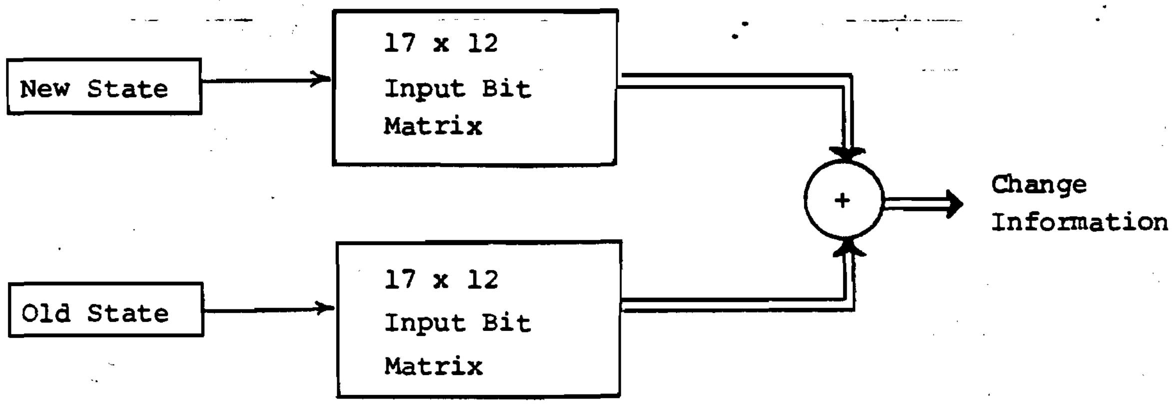


Figure 4-10

Input Data Reduction

words of memory rather than to store the entire state information for each time increment, which would require a matrix of 17 words to be saved for each increment. Continuing on with this first idea, the data reduction was accomplished by taking the first word from each matrix and doing an exclusive-or between them. The result was a word which contained ones in the places were the corresponding bits of the com-

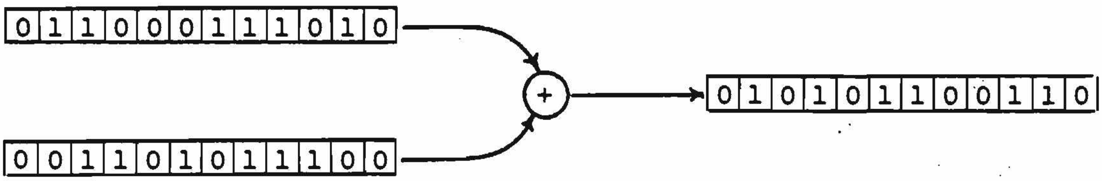


Figure 4-11

Example of the Exclusive-Or Function

pared words differed. If the result was zero (all zeroes in the word), the word was discarded and not saved because there had been no change of state. There are trade-offs again in representing the change information. The trade-offs will be illustrated in the following examples.

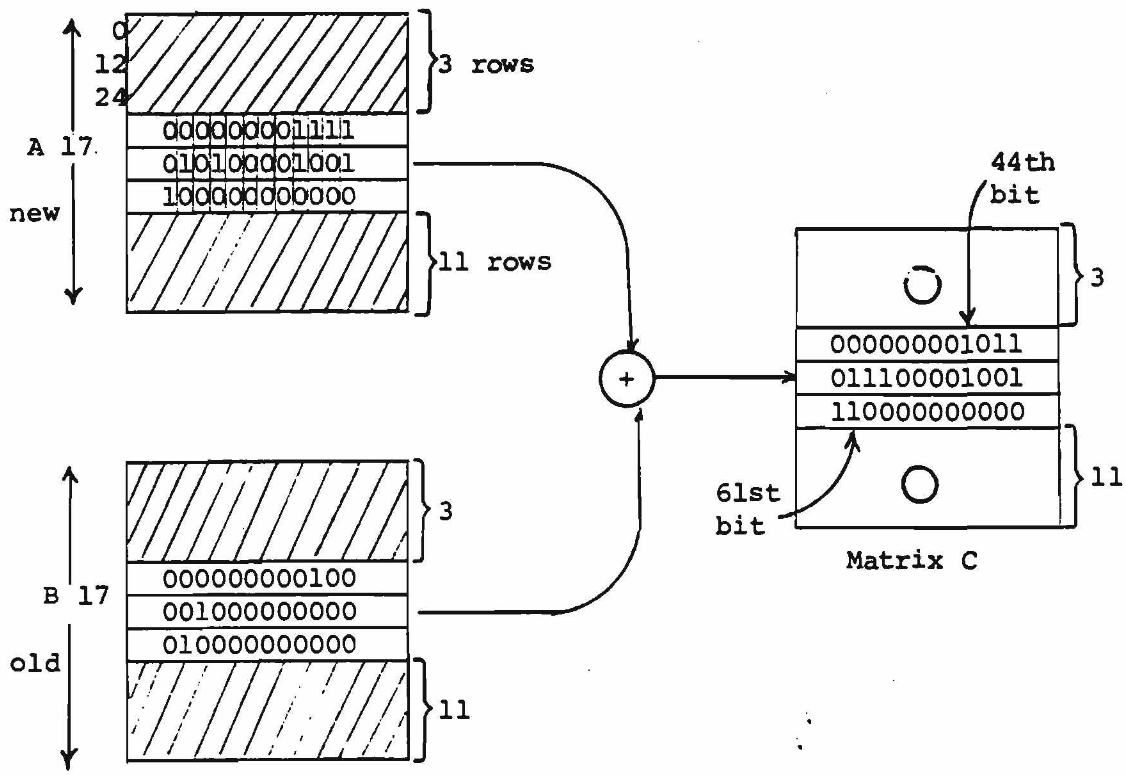


Figure 4-12
Example of Change Information

Suppose the new and old state matrices are identical in the shaded areas. The information we wish to represent is contained in the matrix C. Consider the bits in the matrix to be numbered from left to right starting at the top as shown in matrix A, with an origin of zero. The information we need to save is the position of each change (bits in C). There are 12 x 17 positions, or 204 in the matrix, so an eight-bit number ($2^8 = 256$) is sufficient to designate any position. If the changes were specified by a series of eight-bit numbers, then in our example 8 x 10 = 80 bits of information could represent the C matrix, since there are 10 ones in C: in this case we have reduced the storage

$\boxed{00101100}$	$\boxed{00101110}$	. . .	$\boxed{00111101}$
44	46		61

Figure 4-13

Positional Representation of Matrix C

from 204 bits to 80 bits. This positional representation would be fairly good if we expected only a few randomly appearing bits in our C matrix.

Generally, the changes will tend to cluster in various areas because the keys are mapped consecutively onto the matrix and the fingers of a hand play the notes. The normal playing of chords will produce clusters of ones in the C matrix which signify the depression or release of a key. Taking account for clusters we come up with other representations. There are 17 rows in the example, so a five-bit number may be used to designate a row. One representation would consist of a five-bit number specifying the row followed by the 12 bits of information in the row. In this case, 3 x 17 = 51 bits of information are used to code the information of matrix C:

00100	000000001011	00101	011100001001	00110	110000000000
4		5		6	
row position	data	. . .			

Figure 4-14

Representation of Matrix C

This method of data reduction is good if there are many changes in a few words. Another scheme is a combination of the last two. Here we designate a bit position by an 8-bit number and the following data by a 16-bit vector. The data in matrix C takes 48 bits in this case:

Bit position	
00101100	1011011100001001
41	data
00111100	1100000000000000
60	data

Figure 4-15

Further Data Reduction of Matrix C

If we had decided to use 18 bits of data instead of 16, the representation shown in Figure 4-15 would have reduced to only 26 bits.

The form in which information is going to be used may also influence the type of representation chosen. For instance, on the PDP-8 twelve bit words are handled nicely, and if we want a different unit of information we may easily write routines which will provide us with what we want, but we must pay for it in time. Very often (more often than is done) this time sacrifice should be made in exchange for the data reduction which could logically follow. In practice a hardware mechanism should exist which manages memory and can deliver various sized pieces of data.

Twelve bits of information (PDP-8 word size) is interesting from a musical point of view because the tempered scale of the organ contains 12 notes. The matrices are set up so that rows of bits represent scales beginning with C and ending with B. Transposition of octaves is done by merely logically moving entire rows in the matrices. I will say more about that later. The previous representational scheme with eight positional bits and 16 data bits was based on 24 bits, a multiple of 12. In the example, if there had been no changes in the sixth row of matrix C, the entire information of C could have been represented by the first 24 bits shown in the top row of Figure 4-15. Typically words would be shifted by random amounts in the use of this representational scheme. I am not against the idea of doing a lot of shifting and manipulations, it is simply a matter of our not having nice facilities for shifting; and it turns out that further processing of the information would be easier if we stored data in a different form.

The input data is organized in groups of 12 bit words. The first of each group is divided in half, the first 6 bits specifying row number and the last 6 bits designating the number of following words in the group. In our example the information in matrix C can be represented with 48 bits as is shown in Figure 4-16.

3 data words follow	0 0 0 0 1 1	0 0 0 1 0 0	Start at row 4
	0 0 0 0 0 0	0 0 1 0 1 1	
	0 1 1 1 0 0	0 0 1 0 0 1	Data
	1 1 0 0 0 0	0 0 0 0 0 0	

Figure 4-16

Example Record Data Representation

This representation method has the nice features that it is variable length, preserves the operating unit of the computer, maintains the repeated twelve-note image of the keyboard and tempered scale, and reduces the amount of stored data significantly. This was our initial representation, and after the final processing of the input information, the output information was put in the output matrix which was 33 x 12 bits. The output matrix contained the state of the output switches. Each row in the output matrix corresponded to one of the 33 output address lines, just as the rows of the input matrices corresponded to the 17 input address lines of the interface.

The output matrix was necessary for the conversion of change information to state information. For example, assume that the output matrix defines the 'on' state of several switches. Some information comes that specifies that certain switches should be turned off and others on. It would be nice to be able to feed this change information directly to the corresponding latches and make them toggle, but

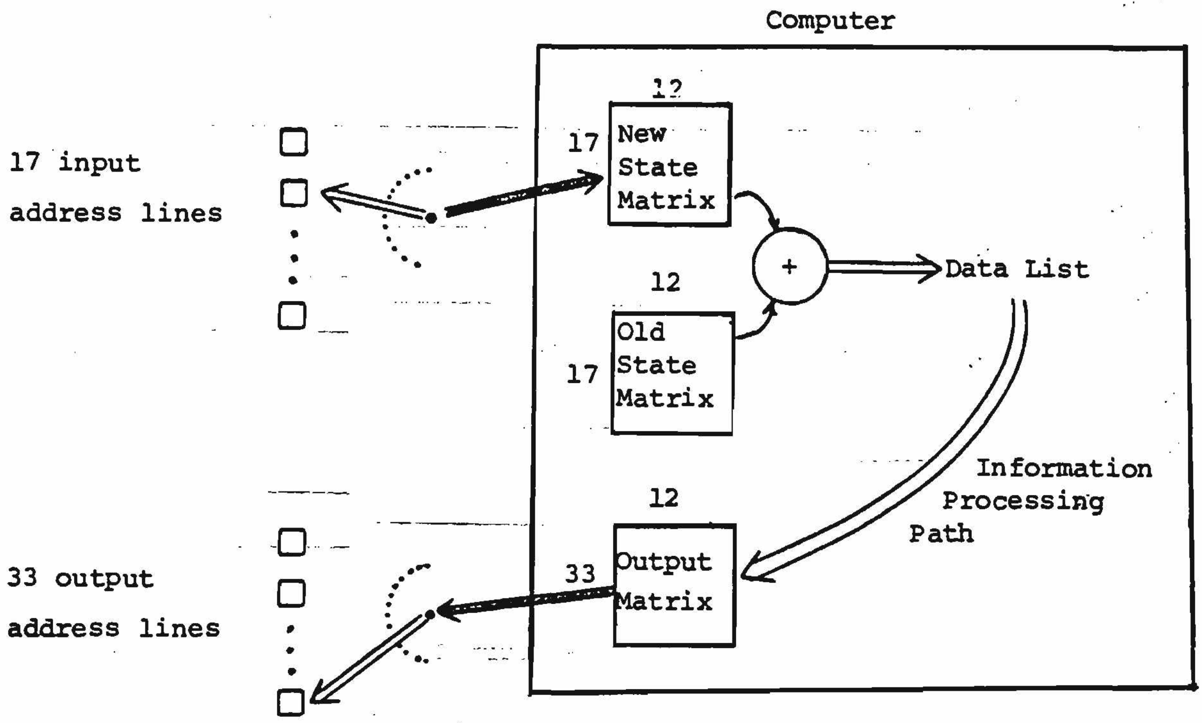


Figure 4-17

Data Flow Using Change Information

our latches do not work that way. Their state, one or zero, must be given to them directly, then they stay in that position until they are given the opposite state signal. The output matrix contained a direct image of the latch states. When output change information came in, it was in list form; i.e., it was represented as shown in Figure 4-16 with a header specifying position and number of words to follow, and some 12-bit data words. The 6-bit starting position word in this case addressed one of the 33 rows in the output matrix instead of referring to one of the rows in the input matrices. Although we only needed 5 bits of positional information for the input matrices, we had to have 6 bits for the output matrix, and that is why the positional pointer was six bits long. The list output information altered the output matrix as follows: suppose the header contained the information $\boxed{j \text{ no. of words, row } i}$. Then the ith row of the matrix was altered by complementing those bits which corresponded to 1's in the 1st data word in the change information. Row $i + 1$ was treated in the same way by the 2nd data word, and so on until j number of words had been changed. The new output state information was then sent out to the corresponding latches. The output matrix preserved the 12-tone scale ordering as did the input matrices, so the direct playback of a performed piece on the keyboard could have been easily accomplished.

The above description was developed on the idea that the data associated with the header word should represent the individual changes as depicted in matrix C (Figure 4-12). Since that time additional work has shown that there is a better way of representing the changing state information. The data reduction is equivalent, but the total amount of storage and processing is reduced. Instead of representing the change information in the data words, the present scheme is to represent the corresponding state information in the data words. In this case only one input matrix is needed, the exclusive-or function is not necessary, and no output matrix is required. The information flow is depicted in Figure 4-18.

The algorithm in Figure 4-18 is as follows: an input instruction is given and the input data word is compared with the corresponding word in the input matrix. (In this case an exclusive-or function does not have to be done since we need only to see if the words are the same.) If they are identical, then the input word is ignored and another input line request is made. If the two compared words are dif-

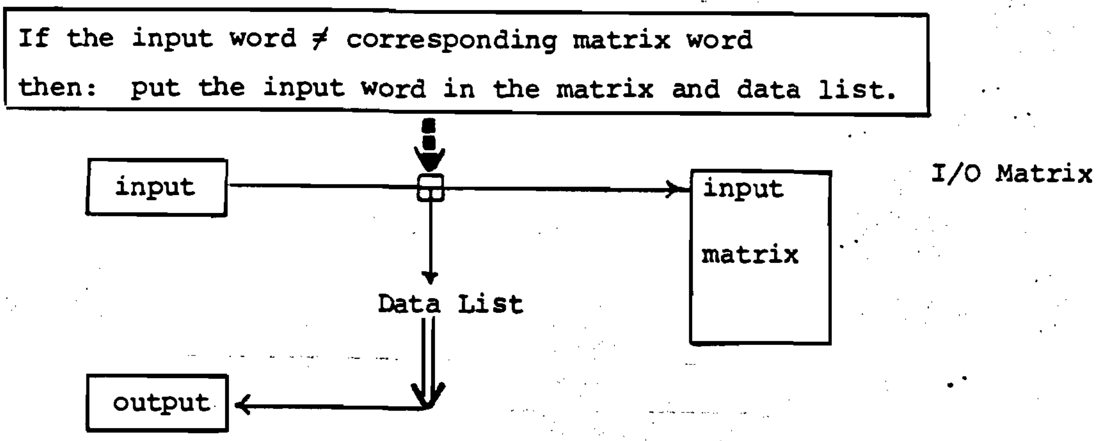


Figure 4-18

Data Flow Using State Information

ferent , then the input word goes both in the matrix and to the data list. Thus the single input matrix takes the place of both the new and old state matrices and the comparison operation replaces the exclusive-or function. Processing speed is gained also becuase the comparison function takes less time to perform than the exclusive-or on the PDP-8. The original scheme with the change information really did not require two input matrices either, since the same input procedure could have been done as was done here, except that the exclusive-or result would have been sent to the data list.

On output, the output list word contains the correct information to be given the latches, so the data is sent directly and there is another significant space and time savings. The output matrix has been dropped because no comparison needs to be done; and the time taken to convert from change to state information has been eliminated. It will still be instructive to think of an output matrix even though it may only exist in condensed versions in the data file. If, for example, a change were to take place in all 33 output words, then the entire matrix would be contained in the output data file, and the representation would be as is shown in Figure 4-19.

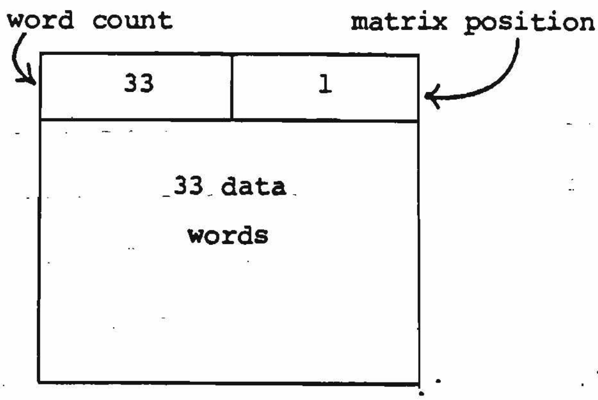
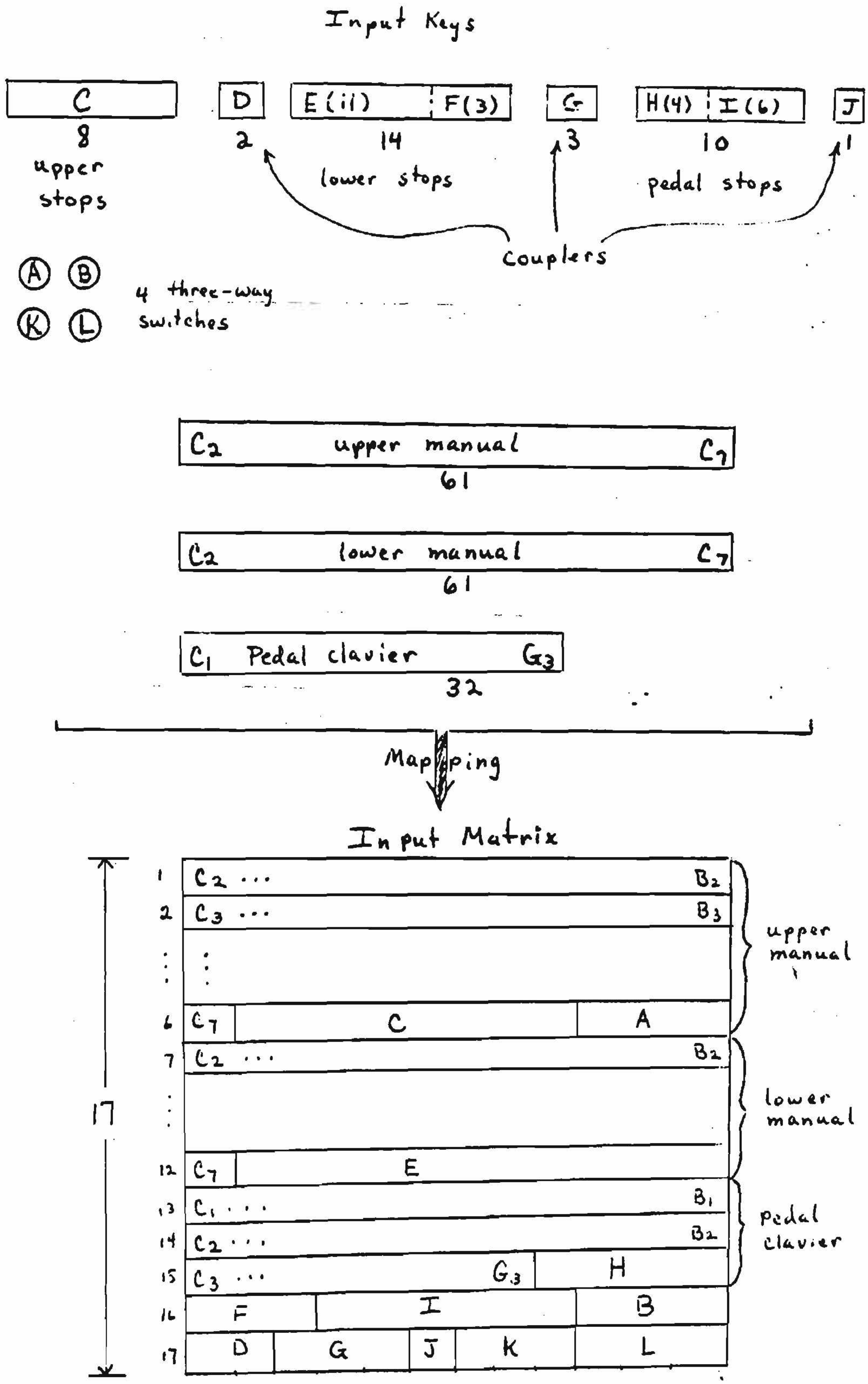


Figure 4-19

File Representation of the Entire Output Matrix

I shall now describe the one-to-one correspondence of the input and (logical) output matrices with the key inputs and switch otuputs. The easiest way to see the input mapping is to show a figure of the organ and the input matrix. (See Figure 4-20)

In summary, the upper manual is mapped into the first 61 bits of the matrix and the rest of word 6 is associated with the 8 upper manual stops followed by the designation of the A three-way switch. The next 6 rows contain the lower manual information followed by the first eleven stop settings for the lower manual (E). Next comes the pedal data in rows 13 through 15 followed by the first four pedal stop settings (H). The final two rows of the matrix contain the rest of the stop settings, coupler and switch position information.



Input Keys
C
8
upper stops
D
2
E(11)
F(3)
14
lower stops
G
3
H(4)
I(6)
10
pedal stops
J
1
Couplers
A B K L
4 three-way switches
C2 upper manual C7
61
C2 lower manual C7
61
C1 Pedal clavier G3
32
Mapping
Input Matrix
17
1 C2 ... B2
2 C3 ... B3
6 C7 C A
upper manual
7 C2 ... B2
12 C7 E
lower manual
13 C1 ... B1
14 C2 ... B2
15 C3 ... G3 H
pedal clavier
16 F I B
17 D G J K L

The output matrix is organized as shown in Figures 4-21 and 4-22 following. There are 85 tone switches on each of the three registration busses, 122 manual lights and 18 filter switches plus an extra switch to make the total of 396.

The output matrix contains the mapped information of the first 84 tones of the three registration busses and the first 60 lights of the two manuals. The five left over C's are mapped in the first part of the 32nd word such that the C_8 of A (Figure 4-21) is in the first bit, C_8 of B is in the second bit and so on with C_7 of the lower light manual in the fifth bit. The rest of the bits in the output matrix correspond to the filter switches. There are not enough filter switch bits in the present version, but we plan to add more filters and expand the number of output lines. The interface addressing is already set up for additional address lines, so the only factors determining the expansion to more filters are time and money.

Data File Structure

A data file is not exactly what the name implies because there are control commands and data represented in a file. Input data may be organized into a data file or be used as control information to determine which data files should be executed. The execution of a data file usually results in the output of data to the organ interface. For instance an input performance can be organized into a file and then the file can be executed to reproduce the performance. It will be interesting in the future to have multiple processes so that, for example, as one process creates a file, another process executes a file. In

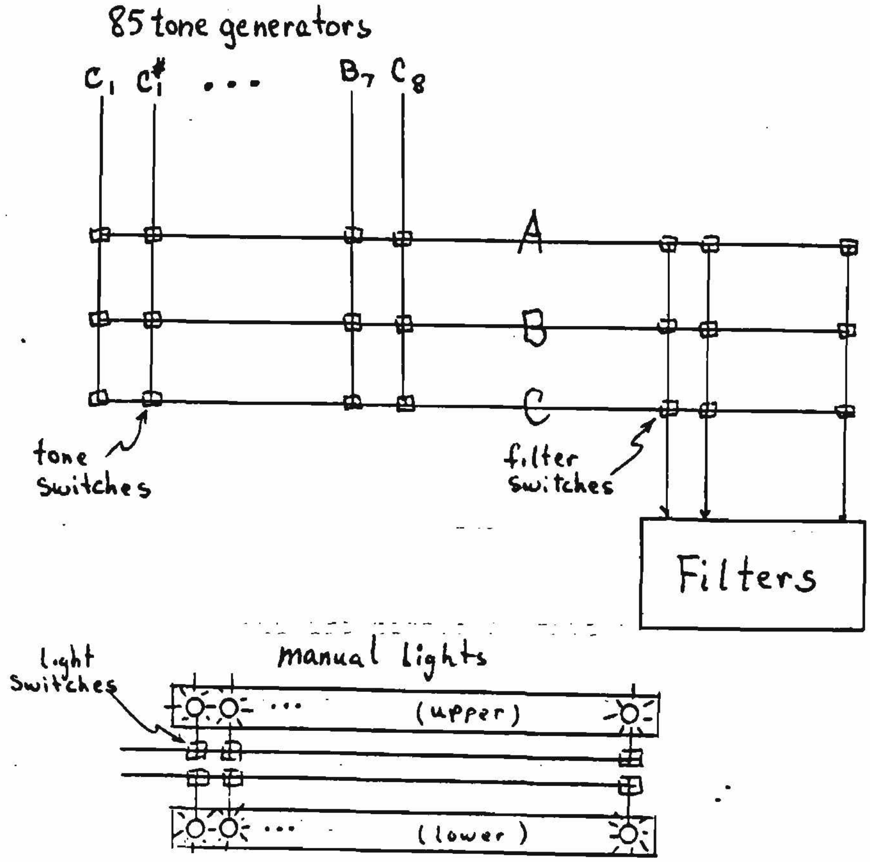


Figure 4-21

Output Switches

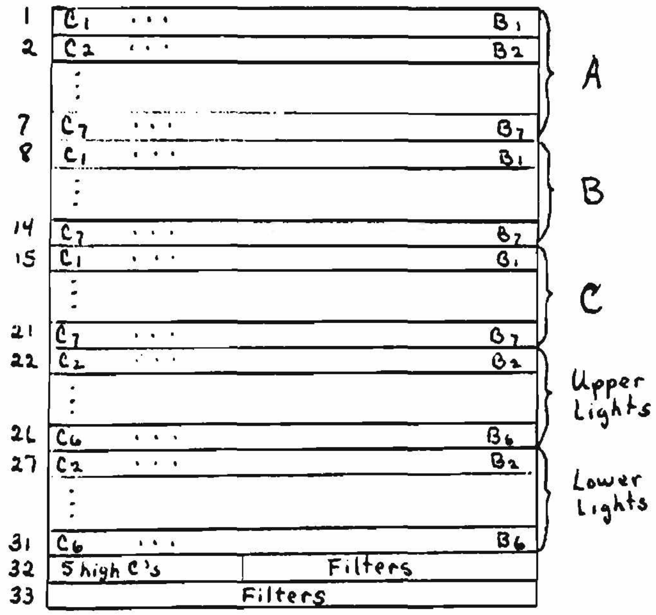


Figure 4-22

Output Matrix Organization

that case the organist would hear the computer-stored version of his performance as he played. Another mode of operation that allows the organ player to hear what he is playing without having the computer execute a data file is the half-duplex mode. It is possible to work in half-duplex by setting the switch for normal organ playing. This switch is also under computer control and thus the operations of half- and full-duplex may be programmed. In the half-duplex mode, the organist hears what he is directly playing and simultaneously a single process can be accepting the input and organizing it into a data file. A data file consists of commands and data. The commands are control information which change the state of interpretation modes. This will become clear as I describe the file structure and the interpreter.

The commands are variable length instructions and there may be an indefinite number of them. That makes it possible to very easily change the data file interpretation by adding additional commands. The plan is to extend the file capabilities to a general purpose "programming language", but the present version is incomplete and must be augmented by PDP-8 machine language programming. The general procedure for extending the file language is outlined below.

There is a hierarchy of commands, the first few of which are four bits in length. (A byte is four bits.) They each have their leading bit zero which distinguishes the byte as a command. Data headers all have leading ones, so that control can be supervised such that data is not mistakenly executed as commands. The first layer of commands are:

Mnemonic	Code	Command
NOP	0000	NO OPERATION
PRFM	0001	NOTE PERFORMANCE IN
IN	0010	NOTE IN
PLAY	0011	NOTE OUT
CALL	0100	SUBROUTINE JUMP
RET	0101	RETURN
REPT	0110	REPEAT
ESC	0111	ESCAPE

Types of data are tagged so that numbers, notes, and addresses, etc., are always identified. The idea is that an argument is treated differently depending on what it is. If the program is asked to play some notes and the data is marked NUMBER, then a warning flag may be raised that says an illegal attempt was made to PLAY something besides NOTES. Addresses will not be treated as numbers, but as locations; when an address appears where a number is expected, the number is sought for at the address. Prespecification of data types, modes of operation, indices, and repeat counts, saves memory space and control time in the long run. Not only is memory conserved, but control is restricted so that programs are easier to get running correctly. The first data types are listed below. I say first because we will surely want to add more types later on; for instance, we might even want floating point numbers at some time for a particular application. Each piece of data is preceded by a header which types the data and contains other various information depending on the type, such as length of the data. Lengths are specified by counts in bytes (4 bits) or in words (12-bit computer words).

Types of Data	Code	Mnemonic	Header Description
NUMBER	1001	NUM	1001 | Byte count | Data Data . . .
ADDRESS	1010	ADR	1010 | Byte count | Data Data . . .
NAME	1011	NAM	1011 | Byte count | Data Data . . .
NOTE BLOCK	1100	NOTE	1100 | NUMBER . . . . Word Count | Matrix Position Data . . .
FILE	1101	FILE	1101 | Word Count File . . .
ESCAPE	1111	ESCP	1111 | New data types Data . . .

The second order of commands includes the following. (We will undoubtedly be increasing the size of this set as time progresses.) The codes of these commands are the next two bytes (8 bits) which follow the escape command, 0111.

Mnemonic	Code	Command
ZOUT	00000001	ZERO ALL OUTPUT
ZIN	00000010	ZERO ALL INPUT
ZIO	00000011	ZERO IN/OUT
STOP	00000100	STOP
DPLX	00000101	CHANGE DUPLEX MODE
ASET	00000110	SET B STOPS
BSET	00000111	SET B STOPS
CSET	00001000	SET C STOPS
AVOL	00001001	A-VOLUME
BVOL	00001010	B-VOLUME

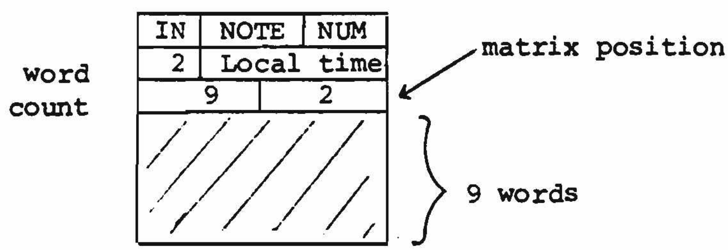
CVOL	00001011	C-VOLUME
TIME	00001100	GLOBAL TIME
	•	
	•	
	•	
ESC	11111111	ESCAPE

Description of the Commands and Data Types

Command	Description and Operation
NOP	The NOP means to ignore this instruction and to go on to the next instruction. It might be used as a place holder for future commands, or it may be the result of a command deletion. When files are compacted to save space, NOP's are simply removed.
PRFM, FILE	The NOTE PERFORMANCE IN instruction takes a FILE argument and builds a data file which corresponds to the organist's input in the FILE specified. The header of the file specifies the length in full computer words, and the data file is built up until the length of the file is exhausted or another command is given which stops the recording process. The constructed file will consist of a series of NOTE BLOCKS with the timing information determined by a real-time clock. Thus the sequence of notes played with corresponding time information is

preserved in the specified file. The routines which accept the input and build the file are presently written in machine language, but later the plans are to be able to control all processes with data file instructions which will operate with operand stacks. If a NAME or ADDRESS parameter appears as the argument, the appropriate addressing will be done until a file header is found, or until illegal data (an instruction or number, for example) is encountered.

IN, NOTE BLOCK The NOTE IN instruction takes one argument—NOTE BLOCK, which specifies the information to be entered from the input matrix. For instance, the following would cause the second to the eighth rows of the input matrix to be placed in the shaded area:



Of course, the NOTE BLOCK could be located somewhere else (not directly following the IN command); and in that case the argument following the command would address the desired NOTE BLOCK. Typically a NAME might follow an IN command. Again, a machine language routine is invoked, as is the case in most of the instructions, to transfer the information from the input to a NOTE BLOCK.

PLAY, NOTE BLOCK — The NOTE OUT command takes a NOTE BLOCK for an argument and sends the information in the NOTE BLOCK to the organ interface. The output address line (which corresponds to the row of the logical output matrix) is determined directly by the matrix position word in the NOTE header. This index is incremented for each succeeding output word until the word count is exhausted.

Actually the NOTE IN and NOTE OUT commands could be combined into a single IN/OUT command that depends on the value of the matrix position pointer as to whether input or output is to be done. For instance, if the position specified in the NOTE header is 17 or less, then input is done. Otherwise, output is designated. It is logically simpler to think of two separate commands in some cases, and programs are a little easier to read.

CALL, ADDRESS — The SUBROUTINE JUMP instruction has an ADDRESS as a parameter. When the CALL is executed the Program Counter is pushed on the return stack along with the STATE WORD and the control is transferred to the address specified by ADDRESS. Suppose CALL ABE were the instruction, where ABE is a NAME. The name ABE would be looked up, or hashed, to yield an address or another name. If another name resulted, the hash scheme would be repeated, etc., until an ADDRESS was found; then the control would be transferred to the address specified by the putting of the address in the program counter.

RET	The RETURN command takes no arguments. It restores the saved program counter and STATE WORD from the return stack and pops it appropriately.
REPT, NUMBER, instruction	The REPEAT command has two arguments, a NUMBER and an instruction. The NUMBER is placed in a repeat count register and the instruction is repeated as determined by NUMBER. REPEATs may be nested through repeated subroutine calls, but not directly. The REPEAT of another REPEAT instruction is not allowed at the present time (it causes an error condition)--although it could easily be made legal if the repeat counter is stored along with the STATE WORD and the return address at a CALL, so nested REPEATs may easily be set up using repeated CALL instructions to routines which themselves contain repeats.
ESC	The ESCAPE specifies that the next two bytes contain the operation code to be interpreted.
ZOUT	The ZERO ALL OUTPUT instruction causes zeroes to be

	put on all the output lines, one right after another, thus turning off all tones, lights, etc.
ZIN	The ZERO ALL INPUT operation zeroes out the input matrix.
ZIO	The ZERO IN/OUT instruction performs both the ZOUT and ZIN functions.
STOP	The STOP function causes the interpreter to cease its execution on the present file. (I envision possible multiple processes, and a particular process is stopped at the STOP command.)
DPLX	The CHANGE DUPLEX MODE switches the mode from half to full duplex, or vice versa, by sending the appropriate information to the last output line. This instruction along with the stop and volume setting commands can be accomplished by PLAY instructions, but it is convenient to have separate commands.
ASET, NUMBER BSET, NUMBER CSET, NUMBER	The SET A,B,C STOP commands have one argument, NUMBER, which is a coded representation of the respective register stop settings. The A, B, and C refer to the three registration busses, each of which can be connected to different filters depending upon the filter switches. The SET commands send pertinent information to the correct output lines to set the switches.

AVOL, NUMBER BVOL, NUMBER CVOL, NUMBER	The A-, B-, C- VOLUME commands operate like the SET commands except that the volume settings on the register busses are determined by NUMBER. Voices, for example, can be set to levels that correspond to the music markings *pp*, *p*, *mp*, *mf*, *f*, or *ff*.
TIME, NUMBER	The GLOBAL TIME instruction sets the clock rate of the main timer. This clock supplies the basic time duration that is counted by the local time information which resides in NOTE BLOCK headers. For example, if the global clock is set to ten clicks per second and a NOTE BLOCK header contains the number five for its timing information, then a half second elapses before the output data is sent to the organ interface. The mere changing of the global time affects the overall tempo of the music controlled and played by the computer.

More commands will be added later, and if need be, the final instruction 1111 1111 can be treated as an escape mode to additional commands. The use of variable length instructions (small data words to express regularly occurring commands and longer words to denote less often used instructions) results in significant data reduction and facilitates complete extensibility for increasingly more special purpose instructions.

Each piece of data is typed by a header so that spurious control can be checked, and so that the type of data can determine the action to be performed. For instance, if a numerical argument is being sought, a NUMBER will cause the immediate data to be accessed; whereas, an ADDRESS will cause an indirection to take place. The data types are described as follows:

Data Header | Description

NUM	nB

NUM (1001) is the code signifying that the data following is a signed integer, nB bytes long, where nB = 0,1 ... 14. The codes NUM and nB are each represented within a byte (4 bits). If nB is zero, then there is no number; this is analogous to a null number. For instance, the representation of a null number followed by a zero is:

NUM	0000	NUM
0001	0000	

where the shaded area is the beginning of other information. If nB is 15 (all ones), then the next two bytes following nB are taken as the length of the number in bytes. In this case, there is a possible representation of 256 numbers, so the length of a number could be 1016 bits long (254 x 4) where the code 255 (1111 1111) is a further escape.

ADR	nB

The code ADR specifies that the nB bytes of data are to be treated as an address. The byte count nB determines the length in the same manner as in a number. If a subroutine call references an ADR, the immediate data is put into the program counter, effecting the jump. If some other instruction references an ADR where a NUMBER, NOTE BLOCK or FILE is expected, the interpreter accesses the information at the designated address and seeks to find the desired argument type. The process is recursive and further indirections may occur.

NAM	nB

The NAM code determines that the nB bytes of data represent a NAME. The byte count operates the same as for NUM and ADR. The data in this case is sent to a hash routine of some kind (it may range from a simple table look-up to a sophisticated algorithm) which delivers an ADDRESS, another NAME or maybe another data type directly. If another NAME results, it will be sent through a hash again, etc.; and if an ADDRESS appears, it will cause appropriate action to take place. For instance, a NAME effects an indirect subroutine jump when a CALL NAME is done and the hashed NAME delivers an ADDRESS. If a NUMBER, NOTE BLOCK or FILE argument is expected, the NAM code will cause an indirection or a series of indirections to take place until the wanted data type appears or until an illegal data type turns up.

NOTE	NUM	nB
(number data)		
nW	i	

NOTE designates that the data comprises a NOTE BLOCK. The words of NOTE-BLOCK data correspond to input and output matrix rows. The code, NOTE, is followed by a NUMBER which represents timing information. This time data specifies a number of global TIME ticks. In the case of NOTE PERFORMANCE IN the time number is set to the number of global clock times that have gone by since the last change of input state. In the case of NOTE IN or NOTE OUT, the time determines the number of clock ticks to wait before executing the actual input or output. A melody to be played by the computer is represented by a series of NOTE BLOCKS with time numbers determining the incremental time intervals between notes. Since the incremental time information is dependent on a clock, the setting of this global TIME rate will affect the over-all accuracy of the input file and the speed of the played music. If a null number is indicated, then the last local time interval is used as a default case. This means that repeated time intervals can be factored out and specified only once. After the NOTE code and the NUMBER, beginning at the next computer words, is the information nW and i. Each of these takes up six bits, and they specify number of data words following and matrix index pointer respectively. In other words, i is the I/O address at which to begin input or output and nW is the number of data words that follow. For instance, in a PLAY command these data words will be sent to consecutive output address lines beginning at the ith address.

FILE	nW

The FILE code specifies that the following nW number of words comprise a data file. A data file contains commands and data which may be executed by the interpreter. A FILE parameter is expected with the PRFM instruction, and the length in the file header determines the amount of storage which can be used by the sampling routine to build the description of dynamic input. The length limit on blocks of storage is important so that bounds are not exceeded unknowingly, causing perhaps disastrous results. For example, the overflow of an expanding input data file could wipe out some of the interpreter, or some other important information. The nW takes 8 bits and represents the numbers 0 to 254. If nW is 255 (1111 1111) then the next 4 bytes contain the word count. The counting begins with the first full computer word which follows the end of the word count.

ESCP	new types

The ESCP code designates that the next two bytes define the data type. The escape allows the creation of an unlimited number of data types. Each time the type code is all ones, twice as many bytes can be used to represent new types.

Examples

1	FILE	33	
2	PRFM	NAM	1
3	A	//////	////
⋮	//////////		
34	ESC	STOP	

200	FILE	200
	200 Storage Words	

Name	Location or Name
A	ADR 200
ABE	ADR 100
B	NAME ABE

Name Table for Examples

Example 1 Input Performance Sampling

In this example a data file of length 33 is defined with the first instruction PRFM. The argument of this instruction is the name A. This name is hashed to yield the address 200 where the file storage resides.

The first line number in Example 2 specifies that the next 6 words is a file. The first instruction is a call to the routine named B. The first hash of B gives the name ABE, (See Name Table in Example 1) and the subsequent hash of ABE yields the address 100, so the control is transferred to the subroutine beginning at location 100. After the

1	FILE	6	
2	CALL	NAME	1
3	B	CALL	ADR
4	2	150	
5	CALL	NAME	3
6	A	B	E
7	STOP		

Routine at location 1

100	PLAY	NOTE	NUM
101	1	0	0
102	4	20	
103	D		
104	A		
105	T		
106	A		
107	PLAY	NOTE	NUM
108	2	25	
109	3	20	
110			
111			
112			
113	RET		

Subroutine at location 100

150	PLAY	NOTE	NUM
151	2	20	
152	2	30	
153			
154			
155	PLAY	NOTE	NUM
156	2	20	
157	2	40	
158			
159			
160	PLAY	NOTE	NUM
161	2	20	
162	3	40	
.			
.			
.			
	PLAY	NOTE	NUM
	2	20	
	5	30	
	RET		

Subroutine at location 150

Example 2: The Calling of Subroutines

subroutine has executed the control returns (from location 113) to the calling routine at line 3. A subroutine at location 150 is called and executed, then another call is made to the subroutine located at 100 (through the name ABE).

The subroutine at 100 plays two NOTE BLOCKS, one four and the other three output words long, both starting at matrix position 20. The first NOTE BLOCK is sounded immediately because the time number is zero, then a delay of 25 clock ticks occurs before the second is played. The return is then performed.

In the subroutine at 150, four consecutive notes are played 20 clock ticks apart. Upon entrance to the subroutine a period of 20 ticks is allowed to go by then the two output words 153 and 154 are sent to the output addresses 30 and 31. After another pause of 20 ticks the two data words at 158 and 159 are played, etc.

REPT	NUM	1
4	PLAY	NOTE
NUM	2	twen-
ty		
2	30	
NOTE	NUM	0
2	40	
NOTE	NUM	0
3	40	
NOTE	NUM	0
5	30	

(cont.)

RET	

Example 3 The use of REPEAT

The file shown in example 3 is equivalent to the subroutine at location 150 in example 2. In example 3 the PLAY command is repeated 4 times by a REPEAT instruction. The local time is specified to be twenty in the first NOTE BLOCK, and this time increment is used for the other three following NOTE BLOCK's because null numbers are denoted. This example demonstrates how factoring out of repeated information reduces storage space requirements.

Extensions

The internal representation and language was designed to be extensible so that a highly integrated system could be built on a common foundation. Different interpretive routines can work on data files; for example, one routine can execute files, another can build and manipulate them (editor) and a third can display selected parts of them (debugger). Since data is interpreted, depending on the header information, a subroutine can be invoked along an access path of certain information. This routine may perform some designated debugging function or it may accomplish a complicated access function, etc. Although interpretation is somewhat slow, the added control capabilities outweigh the speed losses, especially when the utmost speed is not absolutely necessary. Interpreters are typically made up of several routines which are called in some controlled manner. A good programmer with considerable experience will use many small subroutines anyway, since the only way (according to Edsger Dijkstra [36]) to build reliable large systems is to build from small debuggable units. If the time of an interpreter is matched against the time of a program that makes many subroutine calls, then the interpreter doesn't appear so bad. An interpretive system as described consists of many procedures which all share a few fundamental routines, thus these are not duplicated unnecessarily. In typical programming languages, array bounds may be unknowingly exceeded and control can go wild in data, and as a result, the programmer may receive little help from the aborted program error diagnostics. In other words the final specific error condition that determines the fatal error may not give the user much of a clue as to where he made a mistake.

The pre-specification of data, which was mentioned earlier and shown (example 3 above), can save space; in addition, it can save time. Consider the example of a number of data entries being read into the computer until a certain end-marker comes, as opposed to a number being specified indicating the number of data entries. In the former case, each entry must be tested to see if it is the end-marker, whereas in the latter situation the simple decrementing of a counter to zero can signal the completion. The point is that the time spent in repeated testing can be saved by pre-specification.

Variable lengthed operands and the indiscriminate substitution of names, addresses and immediate operands for each other obviates the need for much reprogramming if the interpreter operators (software or hardware procedures) function according to data type. For example, in a given program with FORTRAN if double precision is wanted some number of redeclarations must be made. If triple precision is needed, then the entire program must be rewritten with calls to special arithmetic routines. In the case of an ADD instruction that works on variable sized operands, the program remains the same and the operands expand appropriately, taking only the space required to represent the number. Thus, the program itself does not need to be redone.

In the future the plans are to add a stack and arithmetic, pop and push, selection, conditional and replication operators to the instruction list, to make the command structure complete, so that all processing can be expressed in the language. Multiple lengthed operands of dyadic operators can be handled nicely with three stacks where each one of the three takes turn accepting the result of the operation, using the other two stacks as argument sources [8]. The pop and push instructions govern the organization and placement of data on the stack, and the selection and conditional operations allow sophisticated control to take place. Replication operators provide a further means of factoring out repeated information and reducing storage requirements.

In summary the internal representations of the data and commands are integrated in a file which is interpreted by various routines. The structures are self typed and variable lengthed to facilitate supervised control and extensibility.

CHAPTER 5

USES, CAPABILITY AND EVALUATION OF THE MUSICATIONAL TOOL

The capabilities of the musicational tool are a direct result of the computational powers of the computer which include three main aspects. The first is the stored program idea which is that a sequence of operations can be stored and executed at will; the second is stored data which is that information can be organized, catalogued and later retrieved to be used; and the third main power of a computer is conditional control which is that the instruction executions and the data manipulations can be conditioned on previously defined, or immediate user-controlled conditions.

The small computer in our project has adequate processing power, but limited storage. We will augument the 4K of core with cassette tape recorder storage, with a transfer rate of at least 880 bits per second. Even at this slow transfer rate a significant number of classical organ works can be handled. For instance sample data taken from the following works revealed the average number of changes per second to be about 10. A change is defined as either the start of a note (or chord) or the release of a note (or chord). Thus each change defines a new keyboard state.

Organ Work	Average Number of Keyboard State Changes per Second
Schubert, "Ballet from Rosamunda" Allegretto moderato ♩ = 96 (many 16th notes)	12.5 changes/sec.
Bizet, "Minuet" Tempo de minuet ♩ = 180	9 changes/sec.
Bach, "Praeludium et Fuga" ♩ = 54	7 changes/sec.
(another) "Praeludium et Fuga"	10 changes/sec.
"Fantasia et Fuga" ♩ = 69	11 changes/sec.
Schmidt, "Passamezzo" 12 Variations ♩ = 92	4 changes/sec.
Le Coucou, "Rondo" (all 16th notes) vivo	16 changes/sec.
Bohm, "Calm as the Night" Andante cantabile 𝅗𝅥. = 42	8 changes/sec.

Assuming an average of eighty bits of information for each state sample and an average of 10 samples per second, the information transfer rate is 800 bits per second. The probable transfer capacity of the tape cassette will be around 1200 bits per second so it appears as though this data rate will be sufficient to support music output of considerable complexity. The limited core memory compels us to use core as a buffer and the tape as the storage medium for long files.

I envision the organ-computer communications network as a tool to aid people in the following areas: musical performance, education, composition, score writing, and the study of multisensory perception.

Music Performance

To begin with, a person's keyboard performance can be stored in memory for later playback. The file that is stored corresponds to a sequence of keyboard states according to time. When the information was entered, suppose the organist chose to use both an eight- and sixteen-foot stop. In ordinary organ performance this causes two notes to sound for each key depression; for example, the sixteen-foot stop determines a note an octave lower than the eight foot stop. At the computer recording of the performance, only those notes are recorded which correspond to the actual keys depressed, regardless of which stops or couplers are set. However, the information defining the stop and coupler tablet settings is recorded.

This information can be considered global in a sense, because when a stop setting is made for a particular registration bus, all notes sounded from that bus are directed through the selected filters. Of course, the filters can be switched in and out under program direction. Suppose that a data file has been built up which corresponds to the performance mentioned above with eight- and sixteen-foot stops. Now let us suppose we wish to play back the performance so that it sounds quite like the original. To do this, the appropriate filters are switched in and the file interpreting routine is set such that the correct mapping will be done from the notated key depressions to the desired output notes. The necessity of a mapping in any case is seen from the following figure which depicts the keyboard (61 notes) and the tone generators (85 tones).

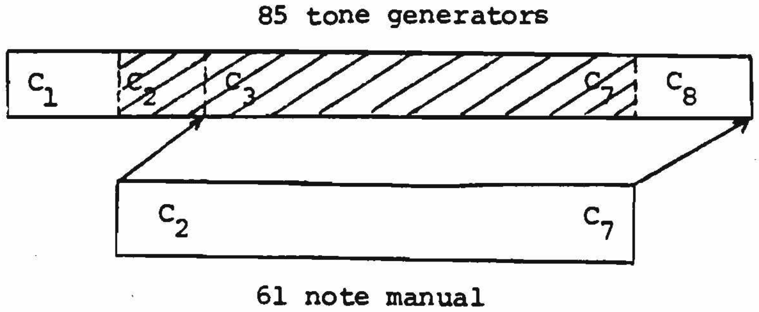


Figure 5-1

Correspondence Between Manuals and Generators

The shaded area in figure 5-1 corresponds to an eight foot mapping of the keys onto the eighty-five tones. The mapping of the keys to the tones C_3 to C_8 shown by the arrows is done by a four foot stop. A sixteen foot stop maps the manual onto the tones C_1 to C_6. With a sixteen-foot stop setting, the depression of the key G_3 would cause the simultaneous sounding of the G_2 and G_3 tones. In our playback example then, the interpreter causes two output words to be sent for each word indicating a recorded key depression. This means that all mappings, indicated by stop settings and coupler selections are done logically by the computer.

An example of a coupler is SWELL TO GREAT 8'. On an organ this means that the voices of the SWELL (upper) manual are transferred to the GREAT (lower) manual keys such that the transferred voices will be heard in their normal pitch range. For example, a SWELL Flute at four foot will be heard at a four foot pitch range from the GREAT keyboard. If, for instance, the SWELL to GREAT 4' coupler were set then the SWELL four foot flute would be heard at a two foot pitch from the GREAT keys. Of course only those pitches will be transferred which correspond to the stops which are set. For example, the SWELL four foot Flute mentioned above would sound from the GREAT keys only if its stop were set. There are also couplers which act intramanually rather than intermanually. The SWELL to SWELL 16' is an example of an intramanual coupler. In this case octave lower pitches would be added to the normal pitches.

The actions performed by couplers can be correspondingly carried out logically by the computer. For example, suppose the upper manual is made to logically correspond to the A registration bus and the lower manual to bus B. Suppose further that a 16' flute stop is set for the A bus (top manual) and a SWELL TO GREAT 2' coupler is to be simulated. The information SWELL TO GREAT (or more appropriately in our case A to B) causes the filters which are connected to the A bus to be connected to the B bus as well. The coupler information (2' in this case) as well as the other registration octave information (16' in our present example) must be noted. Suppose now that a B_3 is depressed on the lower manual and that an eight foot stop is set for this manual. Two notes will be put on the B bus, namely B_3 and B_4. B_3 is a result of the 8' stop and B_4 is a result of a 16' stop being shifted up two octaves by the 2' coupler. This example has been given in detail to show that the ordinary stop and coupler actions can be simulated logically by the musicational tool.

Much more sophisticated mappings can of course be programmed on the computer than the ordinary ones available on organs. There are limitations however to the degree of simulation possible. Consider the case where the SWELL stops, Flute 4' and TRUMPET 8' are set. If this were done right the depression of the SWELL key, say C_3, would cause the C_3 tone to be sent to the trumpet filter and the C_4 tone to the flute filter. What happens in our design (because of lack of switches) is that the tones C_3 and C_4 are gated onto the same bus which is connected to both the flute and trumpet filters.

The unit principle of sharing one bank of tones for each registration bus is used by the computer-organ tool. For instance, suppose that 8' and 4' stops are set on the GREAT manual and the GREAT keys C_3 and C_4 are depressed simultaneously. The tones C_3 and C_4 should sound through the 8' filters and the tones C_4 and C_5 should sound through the 4' filters. Thus C_4 should sound twice. Logically, this could be done in our system if there were enough switches available in the organ interface. The actual response is that the tones C_3, C_4 and C_5 are put on the registration bus, which is connected to the flute filters, thus C_4 is shared and sounds only once.

The output performance capability of the computer-organ is three simultaneous voices each of which can be colored by different filters. The tempo, dynamics and timbre of the voices can be independently controlled by program direction, so these parameters can be varied dynamically by the user. For example, as a certain file is played the interpreter can sample input information to be used as control to determine filter selection, volume and tempo. This is interesting from the standpoint that the person acts as conductor of the performed music. He can for instance, speed up one voice, make another louder, and change the voicing of another. As can be seen, a continuum of control is made available from complete user direction to absolute computer control. The latter can be used to render a pre-recorded or prespecified performance stored in memory.

Education

The possibilities for the organ-computer as a tool in education are limitless. A few examples are mentioned below. A person learning keyboard technique can listen to the music he has just played and diagnose any mistakes he may have made. He can compare the sound of his version of a passage of music to a recorded version by listening to them independently or simultaneously. Very often the mere hearing of how a certain passage sounds helps a person play it. In this case, the passage can be keyed in slowly or actually transcribed and entered symbolically, then played at the desired tempo. Another similar situation is where the person is having difficulty synchronizing his two hands and playing the correct rhythms with each. This person could enter both lines independently and listen to the combined result, then he could have the computer play each of the lines while he added the other. In these practicing and learning sessions the person is not told what to do by the computer, but he uses it as a tool to help him learn a skill.

In a similar manner, the computer can serve as a tool for browsing in a large data base which contains various information about musicology, music theory and music performance. The investigator could use the tool to render say a Bach fugue and at the same time have an explanation of the music appear in written form. Examples both of recorded and written illustrations can be stored.

In addition to the auditory and printed response, the person can also request key position response by the glowing of selected keyboard lights. Ordinary music notation can be displayed on the line drawing scope and this information can be used in conjunction with depressed keys or glowing lights. For instance given a displayed score the person may wish to both hear the tones and see the lights corresponding to the chord he points to by the light pen. On the other hand he may wish to see the chords displayed corresponding to the keys he depresses. These proposed examples are afew of the many interesting investigations which can be carried out with the equipment.

To make the power of the computer available to new investigators, high-level languages and representations must be worked out. For instance, the primitive functions for pursuing the stored information and retrieving desired portions of it must be defined and programmed.

The computer aided instruction (CAI) approach to education is very applicable with the musicational tool. In most mundane CAI programs, a student is lead through a specific path without any choice or direction from the user. The type of approach that seems better is that the user decides what he wants to investigate and the computer adapts some way to his responses. The computer is especially good at helping a person learn skills which can be perfected by repetition.

A study was carried out by an IBM research group [148] which had the following substantial results for a programmed statistics course:

Mode of Instruction	Mean Instruction Time	Midterm Average	Number of People	Mean Review Time
Computer	5.3 hours	94.3%	6	25 min.
Lecture	24 hrs. class 25 hrs. homework	58.4%	8	5.3 hrs.

The computer instruction was carried out in such a way that the problems given to the students depended on their responses to different type problems. If a person missed a certain type of problem, then he was given repeated (not necessarily consecutively) practice on that type of problem; on the other hand, if he performed well on another type of problem he was not tested on that type so often. As a student mastered the various types of problems the computer responded by providing new material and additional problems. In the new problems various review problems were thrown in, and the computer continuously adapted to the users response by providing instruction primarily where the student was having problems. Similar programs can be written to teach music theory, counterpoint, harmony, dictation etc. The advantage of this type of learning for the student is that it progresses at a rate commensurate with the individual's ability to assimilate the concepts being taught.

Responsive environment experiments à *la* Omar Khayyam Moore [112] can be programmed to be used by children to learn concepts of music. The ideas involved here have produced very interesting results. Children at the age of five after being exposed to a responsive educational environment for a couple of years, assume responsibilities like the editor of the newspaper which is sent to the parents of the children. The positional indication of lights, printed text, audio response and graphical pictures may all be used to form an interesting learning environment for children.

An important concept involved with programmed learning is that the student should made aware of what is happening and what the consequence of his actions will be. An example of this occurs in a program which instructs the user in the use of APL instructions. At the beginning the student is told how to use the instructional program, then he is given a choice as to which operations he wishes to learn.

Composition

The interactive nature of the musicational instrument makes it a natural choice to be used as a tool in music composition. A composer can store a desired melodic line, recall it, and try different harmonizations as the melody is played back. Stored melodies may be combined at different speeds, and the resulting combination may be stored for further use. Stochastic methods of computer composition can be stored and the composer can immediately interact with the combinations produced by saving or rejecting them. Thus the computer can render suggestions which may be used by the composer.

If a person wants to write music conforming to some specified rules, he can have programs check his compositions and make sure the conditions are satisfied. The idea of having several stored routines which can perform various functions for the composer is important because it means that when certain repetitive actions occur, such as the harmonization of a note by a specific chord, they can be done automatically. Typical computer analysis of stored music can perhaps also lead the composer to new structural insights which he can apply in organizing his future compositions.

There are some instances in rhythmic performance which sound identical but which may be notated differently. An example is the sounding of a staccato quarter note as opposed to an eighth note followed by an eighth rest. In an interactive environment, if the computer responded with an eighth note but a staccato quarter was really intended, the person can immediately edit the displayed information. Other information besides notes such as slurs, accent marks, volume indicators, etc., can be added by the use of the light pen and editing commands. Another interesting device which we are planning to use to input music notation is a multiple depression stenographic keyboard which is presently being interfaced to the PDP-8.

The next problem to be dealt with is representation. The input will assume some form of linear representation and the computer must store the information in an organized way. Examples of this are the time indexed tables used by Stephen Bauer-Mengelberg, extended versions of the data file described in Chapter 5 and graphic terminal display files.

Several mechanical devices have been proposed for the writing of musical scores. These include Effinger's coded-paper-tape controlled music typewriter, matrix printers, magnetic printing inks, and print wheels [65]. A photographic mechanism was designed with a disc containing cutouts of the symbols through which light was shined on a photographic plate. This device was to be used in conjunction with Bauer-Mengelberg's language, but the cost was determined to be too much and the linear music notation wasn't fully defined, and

Score Writing

The area of musical score writing presents interesting possibilities for the organ-computer combination. The music industry needs a simple automated way of producing music scores. A computer driven system with keyboard input may offer a solution. The main problems involved here are man-machine communication, information representation and design of a programmable output device. The man-machine communication is concerned with how the information is input to the computer and how the computer responds. The ideal environment for this is an interactive situation where the user can monitor the computer responses and manipulate the stored information appropriately. The usual form of music input to computers has been linearized strings of information representing conventional music notation. There is the interesting possibility with a keyboard to enter the notes in by playing them. The computer accepts the input, records the time information, and displays the result on the scope as conventional notation. There are problems associated with this scheme which have to do with the fact that the person playing the music cannot play exactly rhythmically whereas the computer is very precise. For example, if the person plays the ryhthm of a triple dotted eighth note followed by a sixty-fourth note when he is really trying to play a dotted eighthand sixteenth pattern, the computer will interpret the input exactly as played, unless there is some type of compensation or smoothing algorithm applied. The computer could be programmed to round off events to the nearest sixteenth-note interval for example.

apparently still isn't [121]. The mechanical printers all have the limitation of a fixed character set. A brief look at a musical score reveals various sized characters and symbols. A line-drawing display scope is capable of putting up variable sized symbols, and thus it becomes an interesting possibility for music printing. One method to obtain hard copy would be to photograph the displayed score. Tektroniks has recently announed (Spring Joint Computer Conference, May 1970) a hard copy machine that attaches to their storage scope. In a matter of seventeen seconds a reproduction of a displayed picture can be produced on a heat sensitive page. The flicker-free picture on a storage scope would also be pleasant to work with in preparing musical scores.

Study of Multisensory Perception

Human information processing of multisensory stimuli will be studied through the use of experiments performed with the musicational tool. Experimental school programs are showing how multisensory stimulation and involvement are producing accelerated learning of musical concepts. One fifth grade program incorporates the use of an electronic organ, graphs and light panels to teach the principles of notation, pitch relations, orchestral timbres, rhythms, and chord progressions with aural and visual reinforcement [144]. Another grade school program has the children place notes on a staff marked on the floor, and listen for their note to sound as the musical selection is played.

Various experiments can be carried out on our organ-computer equipment to help us understand the role played by multisensory stimulation in music information perception. Variations of the following experiment will be investigated: a child depresses any key and the organ responds with a familiar melody. Then the student is told that only those notes will sound which correspond to the melody just played. He then hits various keys until he has finally played the song. After some time he is told to reproduce the melody with all the keys enabled. Variations to this experiment include the use of light cues, color cues, printed text, and graphical reinforcement. Since the computer controls the responses to the person, various degrees of sensory stimulation and reinforcement are possible. The color generator depicted in Figure 1 (Introduction) displays patterns of colors under computer control. The design of this unit is yet in the future, but it is interesting to speculate how a patch of color could be used to reinforce perception. Can a person learn to play a keyboard instrument by color alone? This might be especially interesting for a deaf person. By repetitious association of certain colors with types of chords (minor, major, augmented, etc.) can a person perceive chords more readily? If he shuts his eyes after some long period of color association can he "see" the color correspondence in his mind? Can a person learn to read conventional music faster if he is given light cues on the keys or name cues under a displayed score? Can very young children be taught melodies with light cues? The field is excitingly open

The future of computers in music is exciting because of control, computational and storage capabilities. Digital controlled oscillators, filters and amplifiers will be used with interesting results as they are precisely controlled by a computer. Education of music in all aspects will be supplemented by computer aided projects and devices. New notations, music languages, and methods of man-machine interaction will be discovered and developed. High level representations of conventional and experimental music must be worked out, and new special purpose devices will be produced. Modern day organs are using more and more computer logic technology, and it may not be long before a small computer comes in the organ cabinet along with the electronic organ. Digital control of analog devices has just begun; the use of a computer to control signal routing boards, mixers and a collection of compatible voltage controlled devices is fascinating and exciting. The computer controlled organ will make definite contributions in the future in the areas of electronic music production, information representation, music education and human information processing.

for interesting investigations. What happens to a pianist if the tones descend as he plays the organ keys "up", (to the right)? Can a person not familiar with a keyboard learn to play certain melodies as fast as one who has played, when the direction of the tones is reversed? How well does a pianist cope with music notation where vertical displacement represents lower sounding pitches instead of increasing tones? Is he not at all affected when both the keyboard is reversed and the music notation also, so that the same visual correspondence exists as usual, but the sense of pitch direction is reversed? How does he fare in this set up with a familiar versus an unfamiliar melody? The answers to these questions may help us understand how different people assimilate information and learn responses.

Studies in binaral hearing have shown that the two ears have separate and almost independent nonlinear effects on the brain [76]. For example, when a certain frequency is put in one ear and the same frequency 180° out of phase is put in the other ear, the tones don't subtractively combine. Two beating tones of close frequencies do not beat when heard isolated in each ear. Either both tones are distinctly perceived, or a compromise pitch is heard. In the previous example of a performer listening simultaneously to his own version of a passage and a previously programmed version, the person could perhaps easily distinguish differences if the voices were separately heard in each ear through earphones.

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